VOL. 2 NO. 10
OCTOBER 1977

MODERN RECORDING

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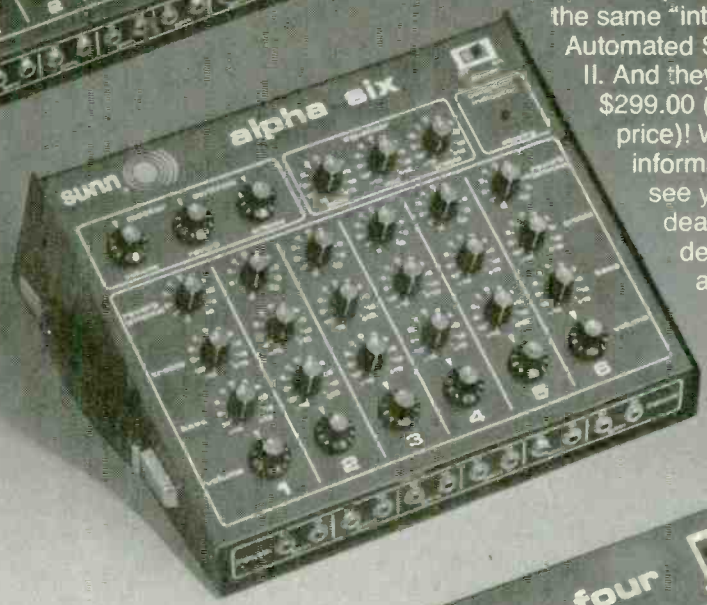

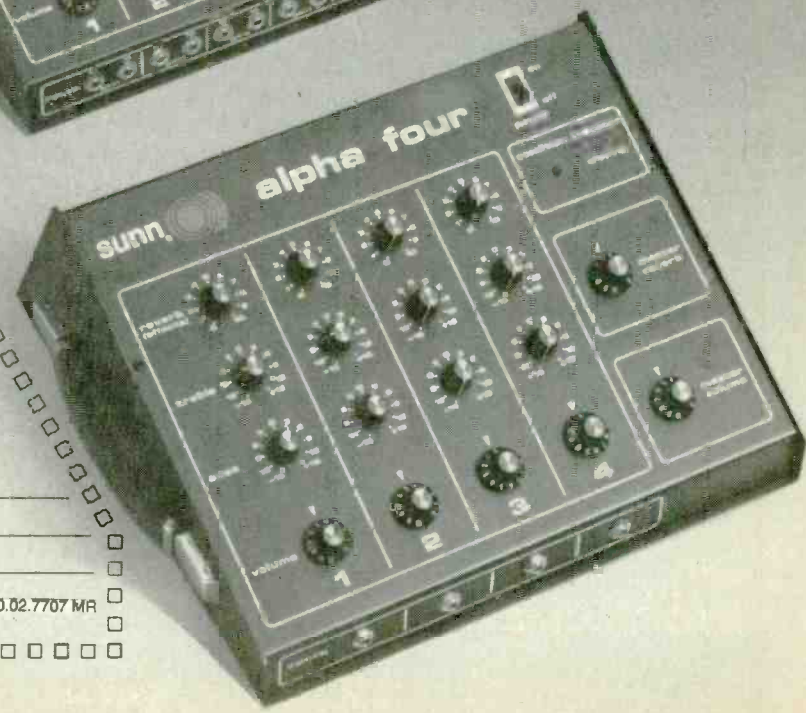

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MODERN RECORDING

OCTOBER 1977
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SERVING TODAY'S MUSIC/RECORDING-CONSCIOUS SOCIETY

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Building a Mobile Studio
Profile: An Interview with Weather Report

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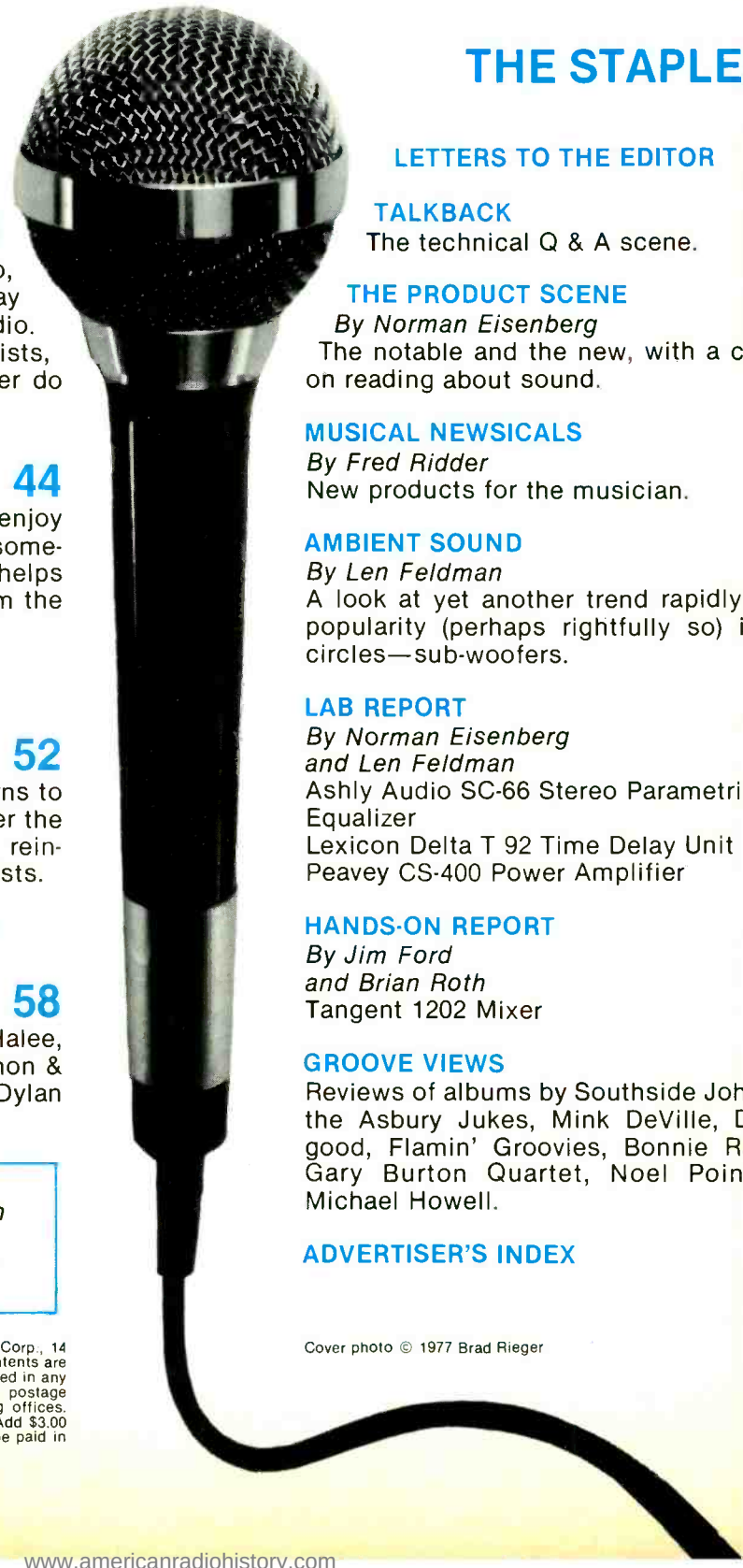

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LETTERS TO THE EDITOR

The World's Smallest Organ?

I really enjoy reading *Modern Recording*. It's a very fine magazine.

I'm eager to find out more about the Stylophone, the world's smallest organ. I know that it is hand-held and played by touching the stylus to the keyboard. How big a range does it have? I have also heard that Paul Anka has composed some of his work on it—is this possible? I've seen ads for it with prices ranging from $19.95 (discount) all the way up to $129.00. Why the vast difference in price? Any information you could pass along on this would be appreciated.

—E. Doty Coy
Richmond, Ky.

By your question, we assume that you're referring to the Stylophone 350-S synthesizer made by Dubreq of London. While we can't tell you whether or not Paul Anka composes on it, we can tell you most of the basics. Just refer to the Musical Newsicals column in the September, 1977, issue. If that doesn't satisfy your curiosity, we suggest that you write directly to Audio Arts, Inc., 5615 Melrose Ave., Hollywood, Ca. 90038.

A Discriminating Reader

I really enjoy your informative book. I think you do a splendid job with Talkback. I don't want to mislead you or for you to think that I am in any way degrading your work by my next statement, but, I enjoy the easy understanding and the simple ways you go about explaining things in the articles. Keep it simple, for many of us are young musicians who really don't know all the ins and outs of studio work yet or what it takes to record something well. I believe it gives musicians a look at what lies ahead and clarifies goals for many of us.

I do have one request: Please don't be afraid to do comparisons on equipment. Some pieces I would like to see compared, for example, are Tascam and Scully boards and tape decks as well as Altec and JBL and Peavey pieces. I feel that the products available today range from excellent to only fair and we should be made aware of the differences. Naturally, money is always a big consideration for there are only certain pieces we can afford.

Thanks for the good work and keep it up.

—Eddy Mattos
"The Sound Company"
San Bernardino, Ca.

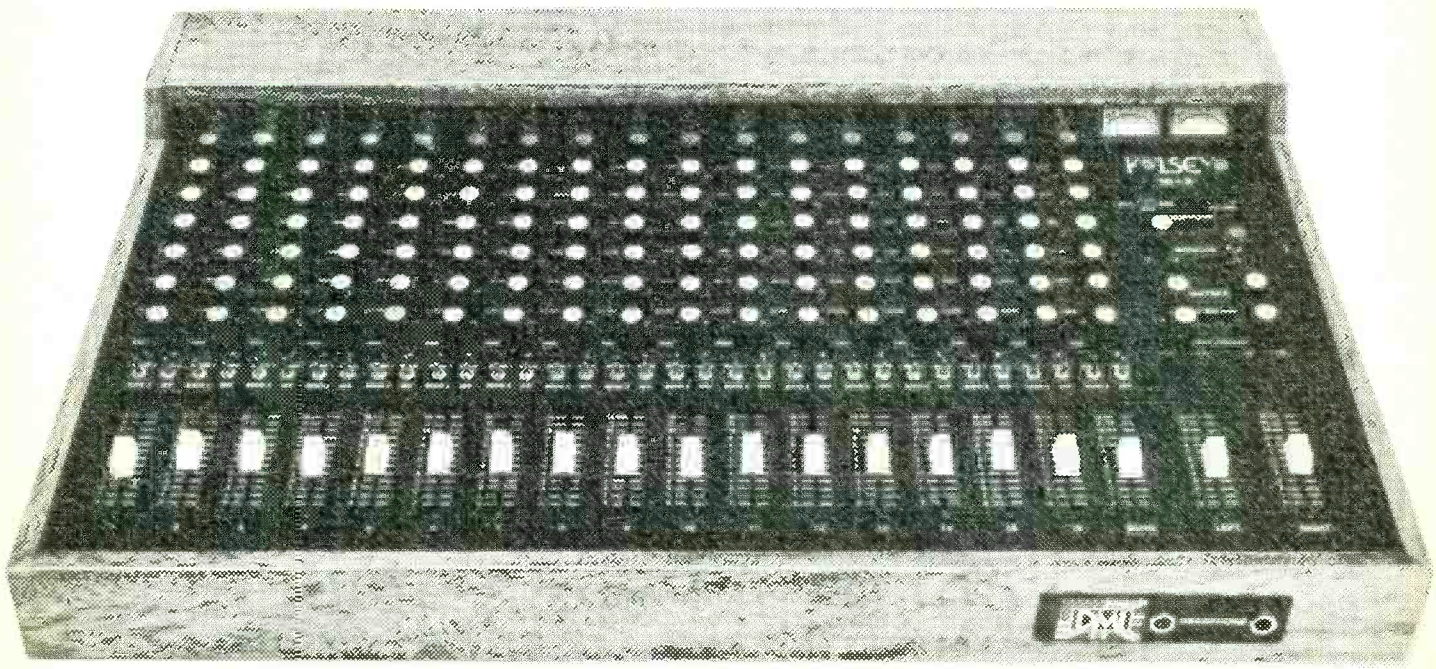
A Classic Recording?

The tone of the letter you are about to read may seem harsh at first, but often, those who believe they are "in the know,"as it were, are, in fact, devoid of the barest essentials of information. I am referring to Russell Shaw's scathing review of the Eagles' release,*Hotel California* (July, 1977, Groove Views, page 85).

There comes a time in the life of a magazine such as yours, when definition of purpose is required. By that I mean, who *really* cares about the lowly opinion of one "record reviewer" with regard to musical quality? In a magazine largely devoted to the operational aspects of recording, personal "cheap shots" at individual musicians are uncalled for.

KELSEY

8, 12 & 16 CHANNEL STEREO MIXERS



If we were to try to tell you about all the features and specifications of our new **Kelsey Stereo P.A. Mixers,** we'd have to take out four-page advertisements. So we're just going to tell you that each input channel has transformer balanced low impedance connector and high impedance jack; gain control; two LED indicators; 3 equalizers; monitor send; 2 effects sends; stereo pan; and on/off/solo switch. And there's an additional effects channel with all the controls of an input channel plus "spin". On the outputs, 2 VUs switchable between main and monitor; left and right faders and tone (high/low) controls; monitor volume; and switchable headphones between solo, main and monitor.

The Mixers have a separate power supply, a solid mahogany cabinet, and come complete in an SMF Tour Series Road Case included in the price. What Price? $4,000.00? No Way!

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12 Channel Stereo	**$1,700 List**
16 Channel Stereo	**$2,100 List**

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Please send me complete information on your Kelsey 8, 12 & 16 Channel Stereo Mixers.

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Whether Mr. Shaw prefers to believe it or not, many people in the music industry and in the buying public found this album to have many highly redeeming qualities, not the least of which is the performance given by "former superstar" Joe Walsh.

There's an old rule among guitarists that goes "knowing when *not* to play is equally important as knowing when to play." Perhaps Mr. Shaw should take this rule to heart in his profession. His comment in reference to Mr. Walsh is not only musically uninformed, but, in my opinion, completely and utterly ludicrous—considering its fantastic sales to date, and the fact that in just a few months, three cuts have become hits.

In ten or twenty years, there will be lots of people around to say "The Eagles' *Hotel California*? A classic recording!" But where exactly will Russell Shaw be? Nowhere, if he continues to take ridiculous "shots" at brilliant albums like *Hotel California*.

I suggest he listen again. And, this time, Russell, keep an open mind.

—Michael Columbus
WLOF Radio
Orlando, Fl.

Areas of Interest

Another letter of praise! I read every word of every issue, underline it as though it were a text, and have learned so much since my introduction to *Modern Recording*.

A couple of areas of interest come to mind, however: I'd love to see an article on Waylon Jennings and Willie Nelson and the Austin music/recording scene.

Also, something on women engineers—since I am one—would be very interesting.

Thanks so much for your fine publication.

—Susan Barrera
San Diego, Ca.

You're in luck—articles on both Willie Nelson and women engineers are looming on the Modern Recording *horizon. They're scheduled for upcoming issues, so keep on reading and enjoying.*

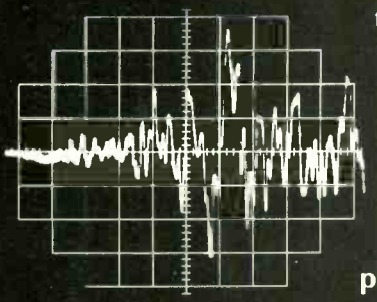
Looking for Employment

I am presently a sound man for a local group and I have taken the Recording Institute of America course. I am interested in obtaining employment in a recording studio, but I have one small problem—I do not know the locations of any recording studios that may be hiring.

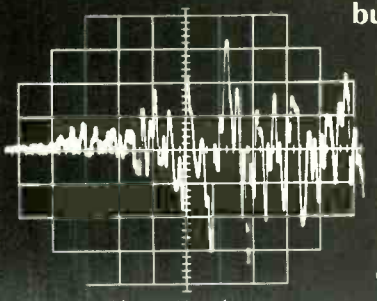
Technics wants you to see what other speakers don't let you hear.

Look at the waveforms. Technics has achieved phase linearity as well as wide frequency response. And that means we've achieved state of the art in high fidelity: virtually a mirror image of music as it was originally played. We call it waveform fidelity. Julian Hirsch, in June *Popular Electronics,* calls the Technics Linear Phase SB-6000A "...one of the better sounding speaker systems we have heard in a long time."

How did we do it? First by conducting exhaustive amplitude/phase studies in acoustically perfect chambers before designing and manufacturing each of the wide frequency/low distortion drivers. Then by developing a unique new phase-controlled crossover network that compensates for the time delays caused by the wide range of frequencies in all music. While simultaneously compensating for the different acoustics of the woofer, midrange and tweeter. And finally by aligning each driver unit in the optimum acoustic position for precise linearity.



Live Piano Waveform.



Piano Waveform reproduced by SB-7000A.

It's a lot of complicated engineering, but it all adds up to something very simple. Music as it was originally played. Nothing more, nothing less. And that's a lot.

Listen to Technics Linear Phase SB-7000A, SB-6000A and SB-5000A. They're now available for demonstration at selected audio dealers for very selective ears.

Technics *Professional Series*
by Panasonic

CIRCLE 66 ON READER SERVICE CARD

If at all possible, I would greatly appreciate the names and locations of studios in the north Jersey area that I could look into.

Thank you for your time and assistance.

—Mark Lorusso
E. Rutherford, N.J.

You must realize that you are dealing with an extremely competitive field, and so our first piece of advice to you is not to be easily discouraged. We feel the simplest thing for you to do is look under the heading, "Recording Studios" in the Yellow Pages for the areas in which you are interested. However, if at all possible, you shouldn't limit yourself to a specific geographical area. Another publication that could prove very helpful is the Billboard International Directory of Recording Studios. This directory is published annually by Billboard Magazine and costs approximately $10.00. You can write to them for a copy at 9000 Sunset Blvd., Los Angeles, Ca. 90069 or call them at 213-273-7040.

Knowledge for Consistent Sound

In response to your article on the Bottom Line, N.Y.C. (MR Dec/Jan 1977) and your intended series on sound reinforcement systems in action, may I say hoo-ray!! The information contained in that article plus the pictorial layout, should be ingested by every serious club owner in the world. Over the past five years, our group has worked with so many sound variables that it has, as you stated, always caused performance degeneration. But with continued information from your sort of intelligent source, broken down so that even the layman can understand, there may finally be places where audiences may enjoy "live" performances with consistent studio quality sound and without a serious loss of hearing.

I could rave for pages on the benefits derived from this info, but let me just say, "Please keep up the series Don Ketteler, whoever you are. You're working for the best professional sound magazine I've ever seen!!!" Continued success!!

—Bruce D. Fielding
Ocean City, N.J.

Info on Engineering

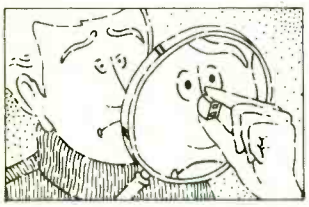
I'm writing this letter with two purposes in mind. First, I have just seen a copy of the Dec/Jan 1976 issue of *Modern Recording* and liked it. I'd like to know which back issues are available and how

ARE YOU BLAMING YOUR TAPE RECORDER FOR PROBLEMS CAUSED BY YOUR TAPES?

Every day people all over the country go into hi fi dealers with complaints about their tape recorders.

When in reality what they should be complaining about is their tapes.

Because the fact is, a lot of the problems that plague tape recorders can be attributed to bad tape.



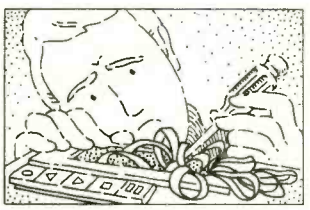
HEAD WEAR IS CAUSED BY YOUR RECORDER. OR IS IT?

If you have to clean your tape heads more than usual, for example, it could be your tape doesn't have a special nonabrasive head cleaner.

Maxell has one.

If your recorder jams, it can be any number of things. Maxell does something to prevent all of them.

We make our cassette shells of high impact polystyrene. And then so they won't crack



JAMMING IS CAUSED BY YOUR RECORDER. OR IS IT?

even after years of use, we finish them to tolerances as much as 60% higher than industry standards.

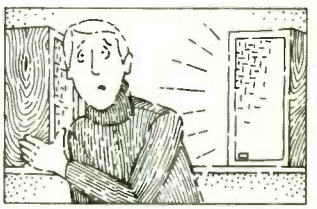
Inside, we use free rolling Delrin rollers so the tape doesn't stick.

And finally, we screw instead of weld everything together because screws make for stronger cassettes.

If your recorder frequently suffers lapses in sound, it could be the tape is of inferior quality. And nobody's bothered testing the tape for dropouts before it leaves the factory.

DROPOUTS ARE CAUSED BY YOUR RECORDER. OR ARE THEY?

Maxell tape is made of only the finest polyesters. And then every



POOR TRACKING IS CAUSED BY YOUR RECORDER. OR IS IT?

step of the way it's checked for even the slightest inconsistencies.

So if you're having problems with your recorder, try a Maxell cassette, 8-track or reel-to-reel tape.

You might find there's really nothing wrong with your tape recorder, just with your tape.

MAXELL THE TAPE THAT'S TOO GOOD FOR MOST EQUIPMENT.

Maxell Corporation of America. 130 West Commercial Ave., Moonachie, New Jersey 07074.

CIRCLE 59 ON READER SERVICE CARD

to order them. Also, what is the cost of a yearly subscription to *MR*?

My second purpose is this: I would like to earn a Bachelor's degree in recording engineering. Since there are no schools that offer this course of study in Puerto Rico, could you recommend a school in the United States, possibly in New York?

Thank you for your help in these matters.

—Ramon Jorge Morales
Hormigueros, Puerto Rico

In answer to your first question, the back issues of MR *that are currently available include: June/July 1976, Oct/Nov 1976, Dec/Jan 1977, Feb/Mar 1977, April 1977, May 1977, June 1977, July 1977 and August 1977. They cost $1.75 per copy and can be had by writing directly to our Subscription Department. Subscription rates are $12.00 for twelve issues and $22.00 for twenty-four issues.*

As for your second question, we don't know of any degree such as the one you describe. However, we do know that Fanshawe College in London, Ontario, Canada, offers an extensive course in recording engineering. Also, we recommend that you write directly to the Recording Institute of America for information about their non-degree course of study which is offered in cities all across the United States. You'll be happy to hear that they are planning to offer the course in Puerto Rico soon. They'll be able to tell you more about that themselves. Good luck!

Compliments for Klabin

May I say first that your magazine is excellent and is probably the best of its kind around. Please keep it going forever! I haven't missed a copy yet.

Secondly, George Klabin's article, "Noise Reduction—Parts I and II" (May and June, 1977) was excellent. I'm looking forward to seeing more of the same.

Thank you and keep those issues coming.

—Peter Moore
Oklahoma, Ok.

Books on Recording

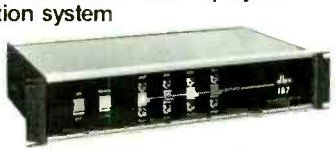
Why not have more "P.A. Primer" articles, including basics on operating sound boards (equalizers and application to various instruments and vocals), unnecessary equipment, helpful additions after basic equipment is acquired, etc.? Also, have more articles with "layman"

dbx noise reduction & signal processing

Tape noise reduction for the professional studio

216 16-channel simultaneous record/playback noise reduction system

187 4-channel switchable record/playback noise reduction system



K9-22 single-channel noise reduction card replacement for Dolby "A"

- 30dB noise reduction with 10dB more headroom
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- Cable assemblies furnished ready to plug in
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- Less than half the cost of a free standing noise reduction system
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157 2-channel simultaneous record/playback noise reduction system

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explanations to recording. Too much is assumed by many writers of your articles when the article is designed for us "dummies" who want to know more.

More books on PA systems and recording would be very helpful if they exist. Maybe MR can find them and advertise. A book review of recording and PA books would be a great help.

—Keith Sims
Fort Worth, Tx.

Your comments are well-taken. We are continually working to get articles of the nature you refer to in MR.

For a listing of books on recording, see MR *issues Oct/Nov 1976, page 24, Dec/Jan 1977, page 18, July 1977, page 6 and the Talkback in this issue entitled, "P.A. Pointers."*

A Request For Results

Have you run any Lab Reports on the following tape recorders and, if so, are reprints of the test results available? The machines I'm interested in are: the TEAC A-650 cassette deck, the Pioneer RT-1011 and RT-1050 open-reel decks and the Pioneer CT-F9191 cassette deck. Any information you can give me would be appreciated.

—Ronald F. Morris
Lakewood, Ca.

Neither the Pioneer RT-1011 nor the RT-1050 open-reel deck has been the subject of a Lab Report yet. The TEAC A-650 cassette deck is slated to be tested by Len Feldman and Norman Eisenberg in an upcoming issue. Some info on the Pioneer CT F-9191 appears in the Talkback section of the August, 1977 issue on page 12. All four pieces appear in the current Modern Recording Buyer's Guide *so you might refer to that for some basic info. Order forms for it are in every issue.*

Amateur Recordists Speak Out

Mind you, I don't normally complain about a magazine's content, but Mr. Passin's response to Hugh West's question of professional vs. "expensive home recorders" for the amateur (June, 1977, Talkback, p. 13) was too much.

Modern Recording so far has seemed to be aimed at the professional and advanced non-professional recordist. I suspect that the majority of its readers fall into one of those two groups, and have minor interest in "copying from a radio program, record or catching his son's or daughter's piano performance."

I think Mr. Passin is out of touch. Today's amateur recordist is likely to be well ahead of the implied beginner's status, and might well have been buying Gotham-supplied microphones for years. After all, our tape decks are sold with no microphones, just like the pros.

—Tom Becker
Binaural Systems Co.
Miami, Fl.

[The following is Mr. Passin's reply.]

I find Mr. Becker's response to my answer of Mr. West's question concerning ". . . home and amateur uses" of Ampex vs. TEAC-type machines to be highly irrational.

If Mr. Becker will re-read the question and my response, he will find that I do not make a distinction between professional and amateur recordists, but between professional and *home* uses of a tape recorder. There can be no real distinction made between professional and amateur requirements and that is why I broadened my answer to the original question.

Our field is analogous to photography. The major differences in cameras are not between pro and amateur but between pro/amateur and the snap-shot cameras. Home recording and snap-shot photography are legitimate, if not critical, uses of their respective mediums.

Mr. Becker will also note that I made no mention of the microphones that come with tape recorders but rather the microphones usually used by home recordists. I think he would agree with me that, compared to the usual microphones used by home recordists, a Neumann KM-84 would certainly improve the results obtained.

—Eli Passin
Vice President
Gotham Audio Corp.
New York, New York

Our Efforts Earn Appreciation

I enjoy your magazine very much and find it includes a veritable plethora of practical, real-life information that the other trade journals seem to neglect. In particular, I have found the works by Jim Ford and Brian Roth very interesting.

Also, John Woram's article in the May, 1977, issue, "John Woram Presents Mike Graphones," makes interesting reading.

Thank you very much for your time—your efforts are appreciated.

—William J. Dickerson
Midland, Mi.

One year later.

Last year, Sound Workshop introduced the 1280; an 8-track recording console of compact functional design that would sound *significantly* better than the rest. Now, one year and thousands of sessions later, the 1280 has become the most respected board in the semi-pro field. Very simply...It sounds better.

The equalizer section of the 1280 provides ±15dB of shelving at 100Hz and 12kHz, and 15dB of peak and dip at 3.7kHz. **Super EQ** was developed for applications where 3 frequencies do not offer enough selectivity.

Super EQ provides 3 bands of equalization with a choice of 5 frequencies per band.

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Low	Mid	High
20Hz	250Hz	4.8kHz
40Hz	500Hz	7.5kHz
90Hz	1.2kHz	9kHz
200Hz	2.4kHz	12kHz
400Hz	4.8kHz	12kkHz (shelving)

The 1280B-8EQ shown with optional Meter Bridge, which works in conjunction with standard **Tri-Lite LEDs.**

1280B (balanced mic inputs)	$3200.
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1280B-12EQ (12 inputs with super eq)	4400.
1280B-EXP (12 input expander)	2500.
1280B-EXP-12EQ (expander with super eq)	3700.
1280A (unbalanced mic inputs)	2850.
1280A-EXP (12 input expander)	2150.
1280 METER BRIDGE (for 1280A and 1280B)	850.

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Advice on Studio Construction

Sometime this year I will be building a house. This will provide the unique opportunity for me to design the house around my recording studio. My questions concern the physical size and location of the studio.

I had planned to put the studio in the basement, on the assumption that this would isolate it from outside noises better than a street level structure. Is this reasoning valid, or do you feel that a spearate building (I could also build a barn on the property) would be a better choice? Let me add that while the studio is in use there would be no problem with household-type noises, such as televisions, children, etc., since the house will only be occupied by two people.

My second question is what would be the ideal size for a studio? The studio would be used to record electronic music and rock bands, sometimes country and western music. (In other words, I see no reason to provide accomodations for the New York Philharmonic.)

If I build the barn, I could make it almost any size within reason. The basement would have some limitations but, even so, its dimensions could be adjusted to approach a more ideal size.

I plan to have a separate control room and a studio large enough for a drum booth, a vocal booth, an upright piano, and the synthesizer equipment (these instruments would be a permanent part of the studio). The equipment would be a Tascam mixer and 8-track recorder plus a considerable amount of peripheral items, such as amplifiers, reverbs, limiters, and so on.

What it boils down to, I suppose, is how wide, how long and how tall? Since the house is in the planning stages, a considerable amount of flexibility is available to me. The basement stairs could be made to enter some portion of the studio, the control room, a separate hall, etc.

Any details you could provide would be of tremendous help to me. I already have definite plans for acoustic treatment of the studio, but no where could I find information on an acceptable minimum/maximum size for a small studio.

Let me add that I get irritated reading all the praise your readers have for *MR*. I feel the space could be better used for more articles. Yet, here I am, telling you that *MR* is fantastic and asking for help. *MR* is a godsend; I genuinely appreciate it, and the only way I can tell you is to write. For years, I'd been hoping for a magazine such as yours, but I thought we were too small a group for anyone to bother with.

Thank you a million times—keep it up.

—Gerry L. Turner
Quincy, Il.

Since the information that you request is too vast to be adequately covered in the space available, we feel the best way to answer your questions is to direct your attention to several articles which have appeared in MR. *"Building Your Own Recording Studio," by Jeff Cooper, which appeared in the Dec/Jan 1976 issue might be a good place for you to start. His article, "How Acoustics Affect Recording," (July, 1977) might also allow you to expand your current plans for the acoustic treatment of your studio-to-be. The book by Mr. Cooper from which these articles were excerpted,* How To Build Your Own Recording Studio, *should be out this fall, so keep your eyes open for that. And, by all means, get in touch with a reputable audio supply dealer near you—he might prove himself invaluable help.*

We hope these help you and—happy building!

We're rather well-known for our expensive amplifiers and tuners.

Now we'd like to be known for our less expensive ones.

When LUX Audio products were introduced to the U.S. in late 1975, we had already played a leading role in Japan electronics for more than five decades. And our audio components had enjoyed an enthusiastic following in Asia and Europe for many years.

Thus, unlike the typical "new" brand, LUX had available a wide array of integrated and power amplifiers, preamplifiers and tuners. Eighteen models in all. Our problem was deciding which to introduce first.

We chose our top of the line power amplifier, the M-6000. Its enthusiastic reception encouraged us to follow with our other state-of-the-art products, such as the L-100 integrated amplifier and the C-1000 preamplifier.

These superb components combined performance, styling and precision in a manner new to the U.S. audiophile. They prompted one audio publication to state: "Almost overnight, the name 'LUX Audio' has earned itself a place of respect in the hi-fi marketplace."

That was fine. Not so fine, however, was our resulting reputation for being very expensive. Thus, many audiophiles were deprived of enjoying LUX quality and performance simply because they were unaware of our less expensive products.

We're now rectifying that. The units shown below are representative of our more moderately priced components. The differences between them and our top models are more a matter of power or flexibility than any variation in quality. You'll see that even our lowest priced units have specifications any manufacturer of fine components would be proud of.

So, if some of the engineering "indulgences" of our top units aren't really necessary to you, you can still enjoy the sonic excellence that distinguishes LUX components... because now you know about our moderate-priced components. Your LUX Audio dealer will be happy to spend as much time as you need helping you to appreciate them.

Luxman T-88V AM/FM stereo tuner. Our least expensive tuner, yet includes FET front end with four-section tuning capacitor. Linear-phase ceramic filters in IF section. Features include FM interstation-noise muting, variable output level control and oscilloscope jacks for viewing multipath distortion, etc. Usable FM sensitivity; 2 microvolts (11.2 dBf) IHF and 2.8 microvolts (14.1 dBf) for 50 dB quieting. Stereo separation: 43 dB at 1,000 Hz. $345.00

Luxman L-80V integrated stereo amplifier. 50 watts minimum continuous power per channel, with total harmonic distortion no more than 0.08 per cent, both channels driven simultaneously into 8 ohms, 20 to 20,000 Hz. Frequency response, 10-50,000 Hz, within 1 dB. Phono S/N better than 77 dB (re: 10 mV input). Bass and treble controls each have 3 turnover frequencies: high and low filters each have 2 cut-off frequencies. $475. (Luxman L-85V, similar except 80 watts per channel, $765.)

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TALK BACK

"Talkback" questions are answered by professional engineers, many of whose names you have probably seen listed on the credits of major pop albums. Their techniques are their own and might very well differ from another's. Thus, an answer in "Talkback" is certainly not necessarily the last word.

We welcome all questions on the subject of recording, although the large volume of questions received precludes our being able to answer them all. If you feel that we are skirting any issues, fire a letter off to the editor right away. "Talkback" is the Modern Recording *reader's technical forum.*

Home-Style Isolation Mount

I am a *MR* devotee in need of information and advice. I recently patched a dbx 124 unit to my Dokorder 1140 and the quality of sound reproduction was truly awesome. My homemade tape was so noiseless that, for the first time, I discovered "sub-sonics" in a real and disconcerting way. I believe that if these rumblings could be controlled, I would have a fine home system. The solution probably involves isolation-type mountings for microphones and for the recording equipment (and anything else that can be set into vibration). Commercially available isolation equipment is scarce and expensive. Can you advise me on material and a methodology for taking up vibrations that is feasible for a home situation? I'd appreciate any practical ideas on this.

—David J. Deutsch
Chicago, Il.

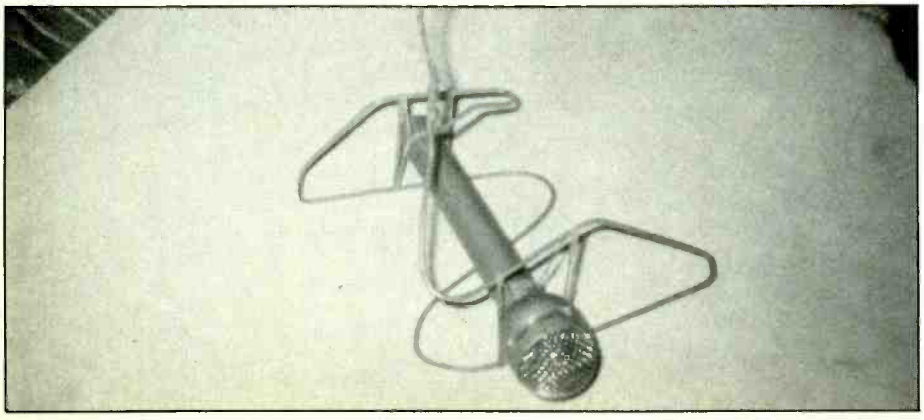
It's doubtful that "shock-mounting" your recording equipment (i.e. mixer and recorder) would help you with your problem but your point is well-taken about shock-mounting the microphone. There's no professional recording engineer that I know of who would be without shock mounts for all of the microphones used.

After a little design consultation I came up with a very cheap yet effective "home style" microphone shock-mount (see photo). Its cost is one coat-hanger and six rubber bands. I was so pleased with the result that I used it on a session the next day!

To build this mount you first bend the coat-hanger into an S-shape with the hook coming out of the side of the "S." Second, put two rubber bands around the mic just behind the head and at the end of the mic case; then thread two rubber bands through the bands around the mic on opposite sides of the mic case. Third, thread the rubber bands over the hanger so that the mic is suspended with two X-shaped rubber mounts; then adjust the rubber bands so that they hold the mic in the middle of the wires and the microphone does not touch the hanger.

This mount can be set on the floor or hung over a chair or other object, bending the hook part of the hanger to aim the mic. If you have a mic stand you can straighten the hook and tape it to the stand.

—Robert Dennis
Engineer
Pro Sound/Superdisc
Detroit, Mi.
Design Consultant: David L. Clark



Omni vs. Directional Mics

I have a question regarding the Hands-On Report by Jim Ford and Brian Roth in the May, 1977, issue which deals with microphones used in professional reinforcement applications ("Microphones for Sound Reinforcement," page 62). Why are omnidirectional microphones seldom used? An omni is more rugged than a directional microphone; it is capable of handling higher levels without distortion; it is quieter; and it will have a smoother, wider range response for the same price. In high-level pickups where feedback and leakage are not problems, the advantages of a good omnidirectional mic would seem to far outweigh the single advantage of the directional unit.

Have I missed something?

—W.J. Dickerson
Midland, Mi.

First of all, this is an excellent question. It represents an area of microphone use that should be investigated by many young sound engineers.

In the "live" situation, there are two items that should make the important difference between the omni and cardioid.

First, feedback and leakage are a big problem. Set up an on-stage experiment, and you will find that the instrument amplifiers and the vocal monitors are so loud you cannot believe it. Many times we have had to do concerts where the vocal monitors are so loud that they start coming through the main front system and begin to color the overall house sound. Most all rock groups will demand very high monitor levels, and if you cannot get it, you are in trouble.

The second important thing is the sound quality. The cardioid mics being used in concert sound reinforcement have a bass boost due to the proximity effect and have high frequency boost due to the design by the manufacturer. These mics were designed this way on purpose, and they give the big "fat" sound that most pop vocalists want.

All of the points that you listed about omni and the cardioid are right. However, microphone selection and use is a matter of particular and individual situations and should be evaluated separately by each engineer. After the technical specifications have been checked, listen to the microphone and let your ears be the final judge.

—James A. Ford
President
Ford Audio & Acoustics, Inc.
Oklahoma City, Ok.

Skew Solution

I am having a problem with my Model 3340-S TEAC tape deck. The tape has a tendency to skew as it passes over the heads. The skew is bad enough to cause the high frequencies to fade in and out during playback. I have adjusted the heads (using the oscilloscope method) and played with the supply reel tension but with no luck. The TEAC company hasn't been able to help. It appears that a tape guide is badly needed between the record and playback heads. Maybe some of your *MR* readers have encountered this problem before since this is a popular machine. Solutions, anyone? By the way, I'm using Scotch 211 recording tape with seven inch reels.

—Bill Stottlemyer
Carroll Recording Studio
Trezevant, Tn.

Things to look for from a repair standpoint are: (1) Position (height) of the left tension arm. It may be too high or low. Bending it slightly either way may solve the problem. I would suggest bending it up, away from the tape-deck face to start with. (2) Insufficient holdback tension on the supply reel, which you have already experimented with. (3) The heads could be mis-aligned. Specifically, one or more heads could be tilted so that all the head and tape guide surfaces are not parallel. If there is any significant wear on the heads, it is unwise to touch the head-alignment screws. It will only make matters worse as the head is probably worn unevenly from being tilted. Even on a head which is originally installed and aligned correctly but shows wear, the chances of proper *realignment* (getting the tape to run properly in the headwear grooves) are slim.

When someone contacts us by phone or letter with a service problem, we can only suggest the obvious fixes as a means of assistance without seeing the unit first-hand ourselves. The problem is not a common one and at this point, we would be interested in seeing the machine at one of our Factory Service Facilities for personal attention. If the owner would contact me at: TEAC Corporation of America, 7733 Telegraph Rd., Montebello, Ca. 90604 (tel. 213-726-0303), I could refer him to a Factory Service representative in his area.

—Tom Spurney
Technical Correspondent
TEAC Corporation of America
Montebello, Ca.

The Lowdown On Mixdown

I am interested in buying the KIK Mixdown Box, but do not know which version to order. I own a Dokorder 7140 4-track multi-sync open-reel tape deck and use Shure high-impedance microphones to record with. Could you please tell me which version would be right for me?

—Alex Davidson, Jr.
Niagara Falls, Ontario, Canada

The key is in the "line output" impedance specification of the tape recorder. If the specification reads "5,000 to 20,000 Ohms," the Model 10K is the right box. 47,000 Ohms, and higher, use the KIK Model 100K.

The KIK Model 10K is designed for use with TEAC, Tascam, Dokorder, and other machines with medium impedance line inputs and line outputs. The KIK Model 10K Mixdown Box is designed for high impedance machines such as Otari. The KIK Dual Impedance Model provides for both types of recorders.

The KIK Mixdown boxes are designed to transfer channels in the following manner:

1) Record the first 3 channels, as outlined in your Dokorder Instruction Manual, either all at once, or using sel-sync for one channel at a time.

2) Return all sync buttons to the "normal" (non-sync) playback position.

3) Connect the 3 line outputs of the recorded tracks to inputs 1-3 of the KIK Mixdown box.

4) Connect the output jack of the Mixdown box to the "line input" jack of the fourth Dokorder channel.

5) Set the "playback level" controls on the first 3 Dokorder channels to "0" VU, or to the balance you want the channels to have.

6) Set the "record level" of channel 4 to "0" VU. (You may want to mix in a "live" mic on channel 4 at this time using the channel 4 mic input, and the mic record level control.) This step can be repeated as many times as necessary

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The astute audiophile is fully aware that, in the speaker business, technological exercises abound. And that many yield marginal improvements at very high cost.

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Rather than esoteric speaker designs for a few, our commitment is to fundamental speaker advancement for many. And that is why B·I·C VENTURI speakers remain way ahead without being way out.

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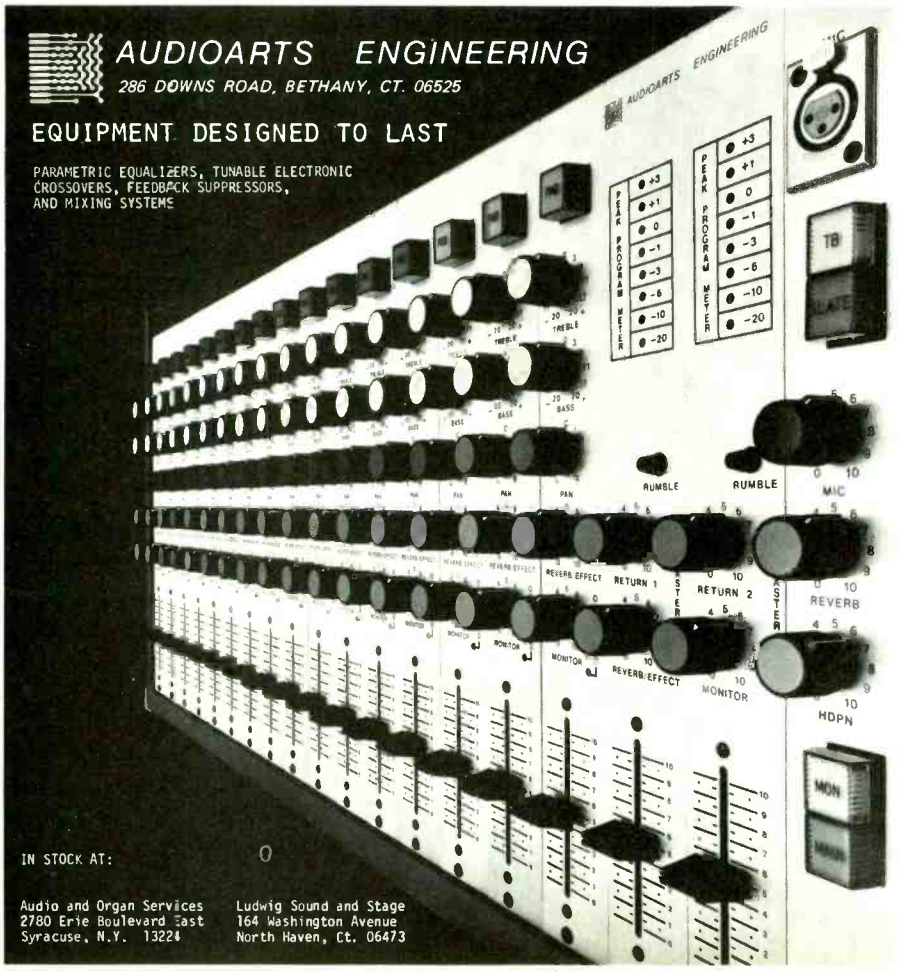

to get a good channel 4 take since the first three channels are still intact.

7) Disconnect the KIK Mixdown Box, and switch Channel 4 to playback in "sel-sync" mode.

8) You can now record over the first 3 channels, combining them with the four channels of information you've summed on channel 4 (using the Mixdown box).

Since the KIK Mixdown box is passive, it cannot generate noise, or distortion, and it gives you the mixing option without the need for investment in a large recording board.

—Harvey Gerst
President
KIK (Hi-Fi Products Div.)
North Hollywood, Ca.

Fighting Feedback

When setting up our P.A. system, I usually try to tailor the sound to the particular hall in which we will be playing. First, I set up the system which includes Cerwin-Vega Speakers, Crown amps and a Tapco board and equalizer. I also set up all mics and instruments before turning on the power. Since we use approximately the same mix every night, the only adjustment needed is the master volume and equalization. My procedure for equalization is what I want to ask about.

I usually set the master volume about where I think it should be set, then turn the output of the equalizer up gradually until the system begins to feed back. As feedback oocurs, I adjust the equalizer to cancel it and continue to boost the volume until I have established about 8-10 dB headroom for the whole system. Of course, I do a little bit of individual eq on the board, but this procedure seems to completely eliminate feedback. Usually the frequencies needing the most adjustment are lower mid-range, between 200 and 500 Hz.

Does this routine seem like it should work? I know we should use a real-time analyzer, but the expense and time involved make that impossible.

Thank you for a great magazine with many great ideas. Keep up the good work.

—Randy Adams
Lindale, Tx.

Although it's difficult to tell from your letter, it appears that there is a flaw in your set-up procedure. Your description seems to indicate that you pre-set the master volume control on the mixer in one spot, then use the level controls on the equalizer for P.A. volume adjust-

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ments. Accepting that the mix and individual eq are set, the level controls on the equalizer should be adjusted for operating level (0 VU) on its meters, in order to take best advantage of its signal-to-noise ratio and headroom characteristics. After you've equalized the system to eliminate feedback by the method you have described, the equalizer gain should again be adjusted to 0 VU. Any further level changes should be made at the mixer's master volume control.

Just as a refresher, there are a few basic rules to follow in order to reduce the possibility of acoustic feedback. Try to place the vocal mics well behind P.A. stacks. Use cardioid or super-cardioid mics wherever possible, especially in critical feedback areas, such as stage front, or near the P.A. stacks or monitors. Restrict the use of omni mics to less critical feedback areas, like stage amp miking. Educate the performers as to how to handle cardioid mics. (Covering the rear vents of a cardioid mic converts it to an unstable omni.) Give roving performers definite physical boundaries for hand-held operation.

If you're using stage monitors, you may find that they cause more feedback problems than the main stacks, due to their close proximity to open mics and high volume levels. Where monitors are involved, mic and monitor placement become as critical as equalization in setting up a smooth, full-frequency, feedback-free sound system.

Without more specifics on your particular system, it's hard to give you advice, but if you try to integrate these basics into your existing procedure, feedback will become less of a problem.

—Bill Mueller
Chief Engineer
Sheffield Recordings, Ltd., Inc.
Timonium, Md.

Isolation and the Distribution Amp

How much isolation can you derive from a distribution amplifier and how much does each output load "down" the amplifier and effect the total output?

—Peter Kouvitsky
East Norwich, N.Y.

A distribution amp is simply a device which will provide energy to a system which "distributes" one signal to more than one "load;" practically speaking, to several "loads." The key to its operation is to have the output impedance of the device so low that changes in "load" effect the output voltage only slightly.

The most common, cheap and dirty application of "distribution amps" that happens around sound studios is the cue amp we use to drive headphones. One input can be fed to anywhere from several to 20 or 30 headphones.If, for safety purposes, you put a 4-ohm, 20-watt or more resister in series with your amp then plug in 0 to 50 headphones, each 4 ohms, the load will vary from 8 ohms for 1, 6 for 2, 5.33 for 3, 4.8/5, 4.2/20, 4.08/50. A very reasonable range for most amps. (I've simply ignored cable capacitance and headphone inductance here.) As you can see,any one headphone that is shorted won't change anything, indeed, if some ballet dancer with cleats shorts out a whole feed, you won't have any change in what's left.

A "real" distribution amp as used in major network production centers, TV stations, some music systems, etc., consists of one input buffer amp which feeds its output to several output amps. This in an even more efficient and

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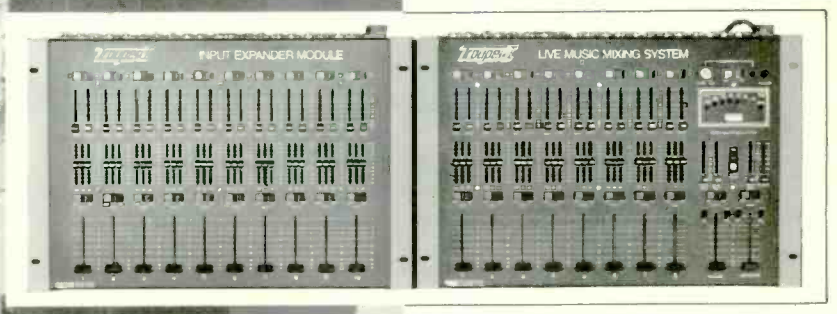

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—Ed Rehm
Chief Engineer
The Nordine Group
Chicago, Il.

Noise Reduction Systems

When mixing down a multitrack tape to the master tape, can a noise reduction or Dolby unit be applied to the master tape with appreciable results on my finished tape?

—C.E. Grier
Montgomery, Al.

Yes! A noise reduction unit may be introduced between the multitrack tape and the two-track master tape to help preserve the quality of the original signal. The two most common noise reduction systems found in professional studios today are the Dolby A system which provides 10 dB of noise reduction below 5 kHz increasing gradually to a maximum of 15 dB at 15 kHz and the dbx system which reduces noise 20-30 dB under what it normally would be. The amount of noise reduction depends on the level of the input signal.

Both systems operate on the compander principle: compression before recording, expansion on playback. Low level signals are raised above the noise level of the tape during recording. On playback, these low level signals are restored to their original level, while the gain reduction lowers the noise level by the same amount. Although both systems use the same principle of operation there are significant differences in circuitry making them incompatible. A noise reduction unit is not a cure for lazy engineering. Good signal levels are important throughout all phases of recording. A poorly recorded multitrack tape cannot be helped a great deal by using noise reduction on the two-track master. Many times noise reduction systems are used on the multitrack tape. I've been involved on projects where the multitrack utilized noise reduction and the two-track master did not and vice versa. It's also common to use noise reduction on both the multitrack and two-track master tape.

Noise reduction systems are useful tools and coupled with good engineering can produce high quality tapes—the result being an enjoyable product.

—Evan Williams
Audio Engineer
United Audio
Santa Ana, Ca.

Only three receivers in the world give you master control of the entire music spectrum.

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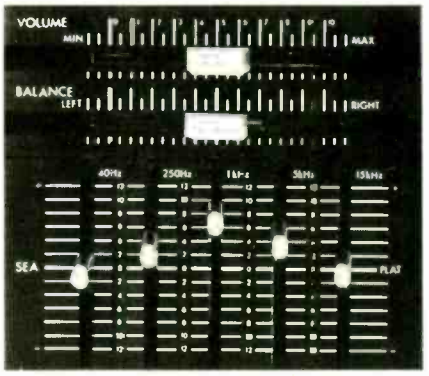
And if you're a recording buff, you'll appreciate another exclusive JVC professional touch. You can switch the SEA equalizer section into the tape recorder circuit for simultaneous equalization while you're recording.

While these unique features alone set JVC's pace-setting receivers apart from the common herd, we're further insuring top performance with a solid combination of additional features.



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So it's up to you. Which do you prefer? A JVC professional receiver with a built-in SEA graphic equalizer. Or one that leaves it out.

JVC America Company, Division of US JVC Corp., 58-75 Queens Midtown Expressway, Maspeth, N.Y. 11378 (212) 476-8300. Canada: JVC Electronics of Canada, Ltd., Scarborough, Ont.

For your nearest JVC dealer call toll-free (outside N.Y.) 800-221-7502.

	S600 II	S400 II	S300 II
RMS power, channel, 8 ohms, 20Hz to 20,000Hz	120 W.	80 W	50 W.
Total harmonic distortion	0.08%	0.08%	0.1%
Approx. retail value	$650.	$530.	$400.

CIRCLE 59 ON READER SERVICE CARD

P.A. Pointers

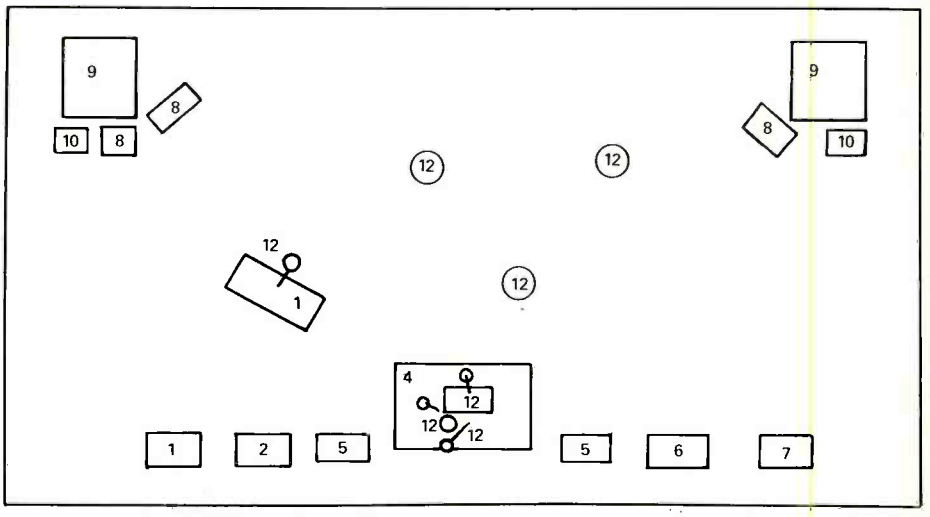
As a guide to inexperienced P.A. owners, could you answer a few questions about the basic problems we continually experience in staging an original rock act? Out equipment set-up is as follows, stage left to right (see drawing):

(1) Rhodes piano, Acoustic 150 w/2-15", 1-12"
(2) Les Paul, Twin reverb
(3) Rogers drum kit, miked
(4) Kick, snare, overhead
(5) Kustom Kasino bass amp, Jazz bass, Hohner
(6) Telecaster, Super reverb
(7) Stratocaster, Twin reverb, 2-12" SRO
(8) Monitor—Peavey PA 400
(9) P.A. Speakers—Sunn Magna 100-2
(10) Power Amps—2 Kustom 8 SRS
(11) Board—Peavey 1200 S
(12) Mics—AKG D1000E

The present monitor set-up is inadequate but necessitated by our financial situation. The P.A. set-up is new and the power amp settings are pure guesswork. The drum kit is miked with three AKG D1000E mics as are the four vocals.

The most common complaints are:

(1) Poor monitor performance. Could this be bettered by different placement of the monitor speakers? What about equalization here?



(2) Fleeting, recurrent high-frequency feedback. Probably due to the number of "live" mics in a tight area and the monitor system.

(3) Recurring ground noise.

(4) General P.A. distortion. Presumably due to overloading the mixer and low settings on the power amps. Instrument amps line into the mixer from external speaker via a low impedance transformer. Bass lines direct to board and to on-stage amps.

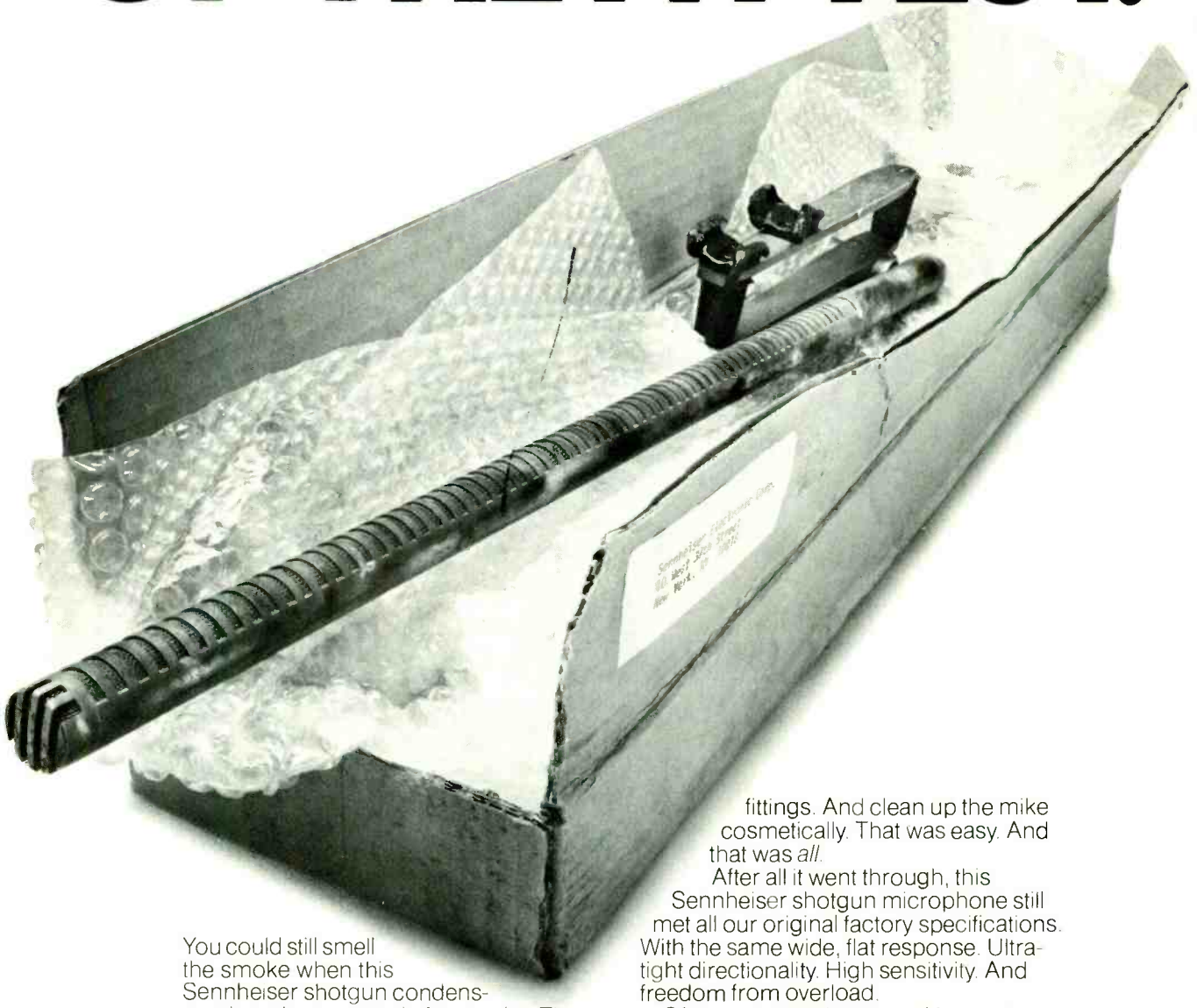
A critique of the equipment set-up and any kind of advice you might find time to impart would be greatly appreciated. The state of the art is very low in Kentucky and there seems to be very few people around to swap gripes with concerning the wide variety of building configurations we encounter.

—Tony Roedig
Fort Thomas, Ky.

There are two points I would like to make before I attempt an answer to a question of this nature. Point one is try to read as many articles, technical papers, and books pertaining to the problems mentioned in this discussion as possible. Point two involves the advice I am offering. The information given below is designed for general use. It may not precisely apply in every case. Readers, please feel free to add to the examples given as the problems listed could obviously go on for pages and pages.

Mic placement, though quite critical, is often difficult to optimize "live." Time of set-up, space limitations and unpredictable room acoustics are your obvious enemies here. Try to keep cardioid mic polar patterns working against feedback. Watch out for hard reflective surfaces particularly behind or above the stage. The temporary hanging of moving pads, or some curtains, aids in control of high frequency reflections. Use good mic technique by singing into mics on a slight angle staying at least one or two inches away. Try using only one mic on two vocals. One mic techniques are often best, contrary to popular belief, particularly when recording. Be careful not to cover vents or ports on mic cases with your hand or with the stand adapter. These holes are part of the mic phase delay network giving it the cardioid pattern. Check mic cords periodically—preventive maintenance is

SURVIVAL OF THE FITTEST.



You could still smell the smoke when this Sennheiser shotgun condenser microphone came in for repairs. The studio where it had been in daily use for years was devastated.

Heat so intense, it melted the shock-mount fittings. Thick smoke, corrosive chemical fumes and water all contributed to the studio's misery.

The studio had no idea how badly the mike was damaged. They couldn't test it. Their equipment had been all but totalled. So they returned the mike to us.

Dubiously, we put the unit on a test bench to see what needed to be done.

We had to replace the shock mount fittings. And clean up the mike cosmetically. That was easy. And that was *all*.

After all it went through, this Sennheiser shotgun microphone still met all our original factory specifications. With the same wide, flat response. Ultra-tight directionality. High sensitivity. And freedom from overload.

Of course, there's no need for such extremes to prove how much you can depend on Sennheiser. Rugged design, precision craftsmanship and the fact that Sennheiser RF condenser microphones don't use high-voltage DC bias (which can cause arcing) have all contributed to our reputation for unsurpassed reliability. And helped make Sennheiser the industry standard over the years.

For more information about our remarkable condenser microphones, see your Sennheiser dealer or contact us. Either way, you'll get a warm reception.

SENNHEISER

ELECTRONIC CORPORATION
10 West 37th Street, New York 10018 (212) 239-0190
Manufacturing Plant: Bissendorf/Hannover, West Germany

CIRCLE 80 ON READER SERVICE CARD

invaluable. Try using phase reverse adapters (Shure A 15 PR) for the equalization adjustments on mics and mic pads. These can all make drastic performance differences. An excellent book for microphone information is *Microphones: Design and Application* by Lou Burroughs. The publisher is Sagamore Publishing Company, Plainview, New York 11803.

I must warn you again, due to the differences in mixing console and effects equipment designs, these answers are general and may not always apply. A call to the manufacturer or experienced users of the equipment in question is a wise investment. However, you might try to run the level controls in a 1/2 to 3/4 open setting for optimum circuit performance. To keep the mic preamps from overload problems, use mic pads. The individual channel level controls will not help if the mic signal is too high. Proper padding should place the level control in the 1/2 to 3/4 position. Use equalization sparingly—large boost increases the chance of clipping and resultant distorted sound output. Try using low frequency cut on any close mic situation. It will reduce rumble and the proximity bass boost inherent in cardioid pattern mics. Beware of hi-fi gear disguised as pro equipment! The equipment I speak of does not have the shielding or balanced line capability of equipment for P.A. use. The time and money spent trying to interface this stuff to your system could be used in purchasing something more useful, perhaps used pro gear.

As for speakers, try running your power amp input attenuators as "wide open" as noise will permit. The outputs of mixers, equalizers and electronic crossovers are not designed to be matched inefficiently. You might also look into bi- or tri-amplified set-ups. Check all crossover points to make sure all drivers are matched correctly. Fuses in the speakers lines can save blown drivers (ask the manufacturer for recommended values). Make sure you use enough power amplification. It is easier for speakers to handle unclipped waveforms than distorted ones from a low power source. Use heavy connectors and check for correct phasing. Use heavy cable on speakers. Number 14 or 16 service cord is a tough, reliable choice. Use balanced lines to the power amp from the mixer (a line transformer 15 K ohms to 600 ohms may be useful). Make sure your mixing console monitor output is not overloading your monitor amp input (distortion here could be causing the squealing feedback mentioned in your letter). Use ingenuity in set-up of your speakers, trying to keep mics and speakers acoustically isolated as possible. Read John Woram's *The Recording Studio Handbook* (Sagamore Publishing). Chapters 4 and 5 can help explain many of the questions related to speakers.

I would like to close by saying again that reading and perhaps some professional, on-the-job assistance are invaluable aids. Here are some other reading materials I personally use and they have helped in many successful solutions: *Audio Cyclopedia* by Dr. Howard M. Tremain, *Modern Recording Techniques* by Robert E. Runstein, free technical literature available from manufacturers is also invaluable as are periodicals such as *Modern Recording*, *Recording Engineer/Producer*, *db*, *the Sound Engineering Magazine*, and *Studio Sound*.

—Carlos Chafin
Audio Engineer
Alpha Audio
Richmond, Va.

When Jerry Garcia, Bob Weir, Steve Miller, Billy Cobham and George Benson all use the AD 230 Delay...

You know it's good!

AD 230

Continuously variable delay up to 600 milliseconds

4 bandwidth selections up to 20 KHz

Built-in flanger with separate controls

Studio quality signal to noise ratio

LED ladder-type VU meters for input and delay levels

High/low impedence with either ¼" or 3-pin connectors

19" rack mount cabinet

And you can bet that these experienced electronic pioneers know how to judge a delay line. The Ibanez Analog Delay with Multi-Flanger does what no other analog device of its kind has been able to do – beat the digital delays at their own game *and* at a price that almost any band can afford. It's unbelievably quiet, features selective bandwidth, and has the most versatile range of controls of any comparable device.

You can get double-tracking, slapback echo, long delay, flanging, automatic vibrato, reverb, and most any other time delay effect possible. Ask about it at your Ibanez dealer today.

AD 220

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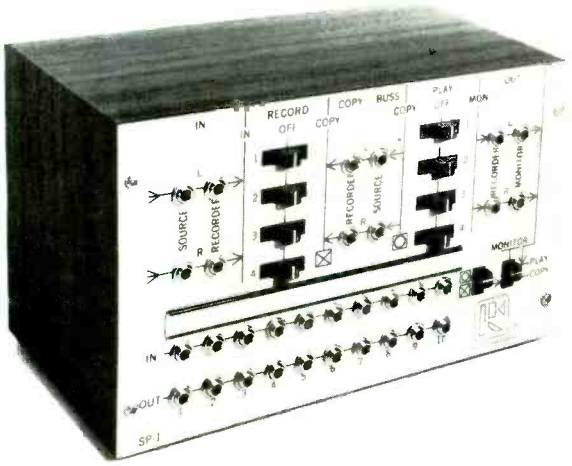
CIRCLE 75 ON READER SERVICE CARD

THE PRODUCT SCENE

By Norman Eisenberg

VERSATILE INTERFACE DEVICE

A new audio patching center, the model SP-1 from Russound/FMP, Inc. of North Berwick, Maine, allows the audio-equipment owner to connect all stereo components and accessory devices together, and to switch them in what is said to be a "virtually

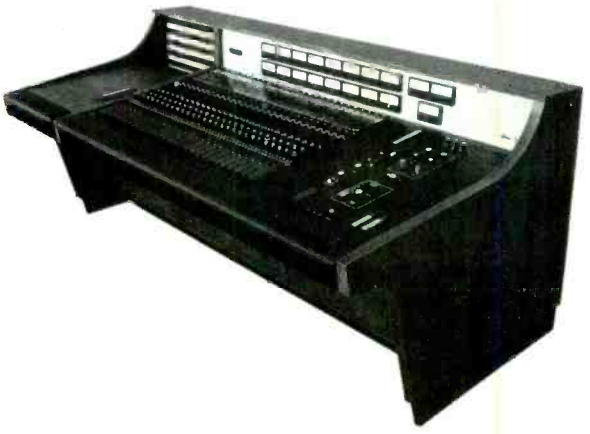


infinite variety of configurations, limited only by imagination and the number of available patch cords." The SP-1 connects to the tape monitor input and output jacks and has 48 additional jacks of its own. Four tape recorders plus other gear can be jacked in at the rear of the SP-1, and their interfacing controlled by front-panel switching or by changing patch cords. Applications listed for the SP-1 include dubbing and copying, decoding from one noise-reduction format and encoding into another, duplicating tapes on several recorders simultaneously, intermixing of several two-channel sources, reversing channels, and many more. Supplied in a walnut finish vinyl case, with twelve patch cords and a twenty-two page booklet, the SP-1 patching center is priced at $149.95.

CIRCLE 7 ON READER SERVICE CARD

SPHERE'S ECLIPSE CONSOLES

Several new consoles in the Eclipse series, designed specifically for multi-channel music recording and available as 4,8,16,24 and 32 track systems with up to 40 inputs, have been announced by Sphere Electronics of Chatsworth, California. Of modular construction and easily expandable, these consoles are claimed to offer very high performance through the use of discrete operational amplifiers and specially designed transformers. And while they are capable of handling complex mixing patterns, the consoles are said to have been designed with special emphasis placed on the "human engineering" of controls to simplify their operation and minimize user fatigue. Among the Eclipse series features are: wide choice of equalizers, all of which are interchangeable, including two styles of graphic; solo in stereo position with echo; solo of monitor channels; monitor mute; stereo echo send; monitor echo; two pannable effects returns; two programmable mutes



on each input; stereo cue. Patching uses professional size ¼-inch jacks. Channel selectors are color coded and illuminated. Optional are light beam level displays.

CIRCLE 2 ON READER SERVICE CARD

SIX-IN MIXER

From Spectra Sonics of Ogden, Utah comes news of its model 1100 Line/Microphone Audio Mixer which accepts six line or six microphone inputs and has a monophonic output. The device also features a monitor capability, high-and low-frequency EQ and a VU meter. Inputs and program output are trans-

former-isolated. Priced at $800, the model 1100 may be installed in a standard rack mount. Among its important specs are: S/N ratio of 80 dB minimum for line input, and 78 dB ± 1 dB for mic input; maximum continuous sine-wave power of +24 dBm ± 0.5 dB; maximum and typical THD rated at 0.02 and 0.01 percent respectively; and equalization capability of ±20 dB at 20 Hz and 20 kHz, and ±14 dB at 100 Hz and 10 kHz. CIRCLE 3 ON READER SERVICE CARD

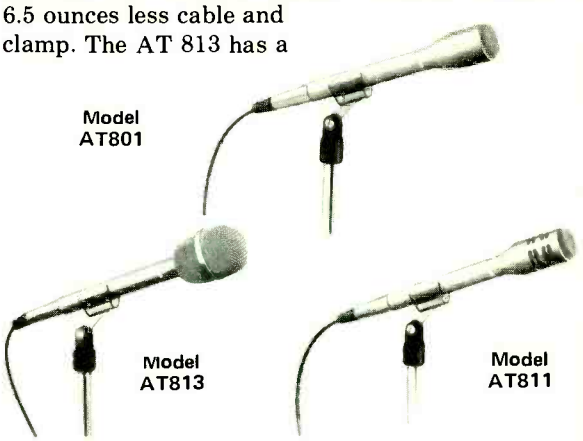
SUPER AMP FROM SOUNDCRAFTSMEN

A specially developed circuit, said to anticipate power demands and then supply only a proportional amount of power as required by varying input signal voltages, is featured in Soundcraftsmen's new stereo "super-amp," the model MA 5002. The amp is rated to deliver 250 watts (rms) per channel, 20 Hz to 20 kHz, both channels driven into 8-ohm loads, with less than 0.1% THD. Noise is listed as better than 105 dB down, and input sensitivity for full-rated output is 1.28 volts. Slew rate is stated as better than 25; damping factor, greater than 100; response, as within 0.25 dB from 20 Hz to 20 kHz. The "vari-proportional" system, as it is called, through its continuous monitoring of output power needed, reportedly results in energy savings by reducing the amount of energy dissipated as heat loss. Yet controlled full power always is available, standing by, and supplied as needed. According to the manufacturer, this design enables a higher-powered amplifier to be sold at a lower price than class AB amplifiers since certain costs are eliminated—for instance the MA 5002 is said to need no ventilating fan even under the most severe operating conditions. Equipped with LED indicators, dual power cords for remote AC on/off, wood side panels that may be removed for rack-mounting, and switching for two speaker systems, the MA 5002 is priced at $699.

CIRCLE 10 ON READER SERVICE CARD

NEW LINE OF MICROPHONES

Audio-Technica has added microphones to its product offerings, which up to now have included phono cartridges, headphones and record-care accessories. The new A-T mics are available in three electret-condenser and two dynamic (moving-coil) types, and each type can be had with cardioid (unidirectional) and omnidirectional patterns. Top model is the $80 AT 813, an electret unidirectional with response rated from 20 Hz to 20 kHz, −58 dB sensitivity, and balanced 600-ohm output. Weight is 6.5 ounces less cable and clamp. The AT 813 has a



built-in protective screen with multistage filters to reduce wind noise and "pops" from close-up use.

Others in the line are the AT 811 at $75, and the AT 801 listing for $50. The AT801 is an omnidirectional type.

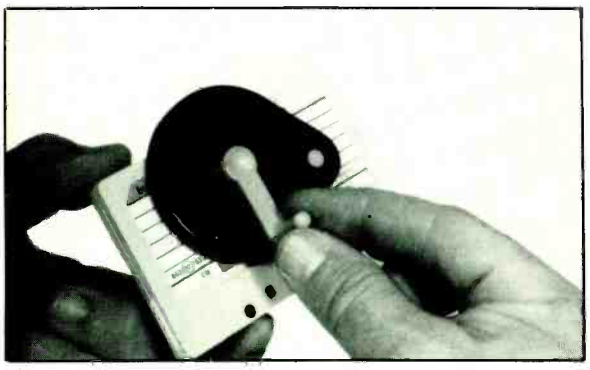
All electret models have permanently polarized diaphragms requiring no external power source. One AA-size cell is placed inside the mic to power the impedance-matching network.

The models AT 802 and AT 812 are the dynamic mics, omnidirectional and unidirectional, and priced at $50 and at $75, respectively. Accessories in the mic line include cables with XLR-3 connectors, snap-on stand clamps, protective carrying cases, line matching transformer, slip-on foam windscreens and a shock mount.

CIRCLE 20 ON READER SERVICE CARD

CASSETTE TAPE WINDER

Robins Industries, the accessories manufacturer, has announced a new high-speed hand-held winder for cassette tapes. Known as catalog no. 36-006, and priced at $4.59, the device permits wind or rewind of a cassette off the deck at twice the normal rewind speed of the recorder. To use the winder, you slip it onto a cassette and lock it in place. A hand-crank operates a gear that drives a keyed spindle which fits the cassette hub.



CIRCLE 16 ON READER SERVICE CARD

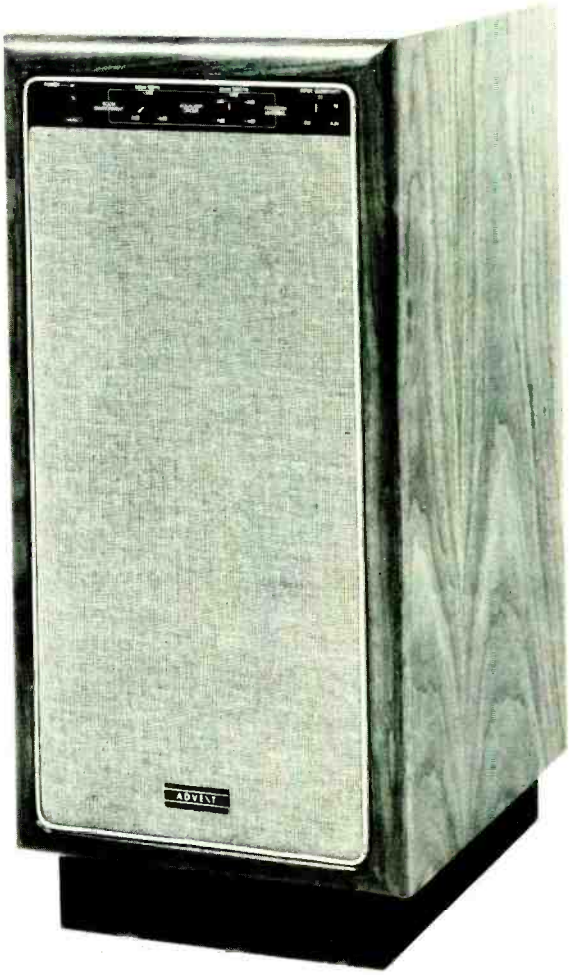
A&D PARAMETRIC EQUALIZER

The model E950 equalizer, from the British firm Audio & Design Recording Ltd., is called by its manufacturer a "paragraphic" device which is their trademark for parametric. In the E950, pre-set, color-coded positions for frequency and bandwidth enable the device to be set up in either stereo 6-section format, or as a 12-section mono format, with output on either of the unit's connectors. In the 12-section mode, the device can serve as an octave equalizer with two "floating" sections for use over the full bandwidth for "tight notching." According to the manufacturer, the E950 should be ideal for sound-reinforcement in tight-notching up to six feedback modes in stereo, removing less audio content than with third-octave systems, and also capable of being exactly tuneable to required frequencies. A&D Ltd. also indicates the device's applications extend to broadcast use and to the recording or sound-processing studio. Frequency is variable over a four-octave range for each section; and bandwidth may be varied down to one-eighth octave. Inputs and outputs are balanced, and the device is rated to handle overload in excess of +24 dBm. Dimensions are standard rack-mount.

CIRCLE 12 ON READER SERVICE CARD

ADVENT OFFERS POWERED SPEAKER

A surprise product from Advent Corp. is its new powered loudspeaker, consisting of a two-way acoustic-suspension speaker driven by an integrated bi-amplification system. Known as The Powered Advent Loudspeaker, the new product is



said to be the result of Advent's on-going investigation of the complex interaction between power amplifiers and drivers and the listening room. "Only by integrating amplifiers and speaker drivers," says Advent, "could these interactions be predicted and controlled."

Priced at $500, the new speaker/amp system is somewhat larger than the conventional 2-cubic-foot a/s cabinet, measuring 28⅜ by 14⅛ by 11⅞ inches. For additional info, write to Advent at 195 Albany St., Cambridge, Ma. 02139.

CIRCLE 1 ON READER SERVICE CARD

"FANTASTIC" PRODUCTS

From the company that calls itself Fantastic Sounds, Inc. (Long Beach, Ca.) comes word of several new products. One is a microphone talk-back system, available in many configurations. It permits switching one mic signal to two inputs remotely via a switch located at the microphone. Thus, for instance, a performer can switch his mic off the house sound system and into a cue monitor, or board-operator headphone in order to communicate with his technical crew during a performance without the audience being aware of the signal-switching. Optional hardware permits the system to remotely switch other types of equipment such as lighting, cue lamps or peripheral signal- processing equipment.

Fantastic also has a dynamic equalizer offering up to 15 dB of boost or cut, in frequency ranges from 50 Hz to 1 kHz, and from 500 Hz to 10 kHz. It may be used as a simple variable frequency equalizer or—in its "dynamic" mode—to vary the EQ from flat response to the selected EQ settings automatically as the dynamics of the input signal change. The threshold at which the EQ changes begin is also variable. Suggested application is for tailoring a sound system's response to correct for a vocalist whose tonal qualities vary as his or her dynamics vary.

Another Fantastic product is a "Tri-State LED Signal Indicator" which is said to be capable of displaying both normal and overload signal conditions; it employs what Fantastic describes as "sophisticated pulse-stretching circuitry techniques" that allow the LED to show instantaneous overload not normally visible on an LED.

Finally, there's a Power Indicator module that allows continuous monitoring of bipolar power supply voltages. In this device, two LED indicators remain on continuously unless operating voltages drop below a predetermined threshold level.

CIRCLE 5 ON READER SERVICE CARD

READING ABOUT SOUND

Sometimes reading about sound can be almost as fascinating as hearing it, and a number of manufacturers know this and take seriously the matter of "literature" offered to prospective buyers. Among the spate of printed offerings that comes across my desk, I often find some of special interest and would like to recommend these recent examples.

Crown International, 1718 W. Mishawaka Road, Elkhart, Indiana 46514, has a four-page brochure listing "preliminary specifications" for its D-440 stereo power amp, and DL-400 stereo control system which involves a three-module design. This is the one you may have heard about—it shows digital readout of each channel. Crown also has released an article on equalization that is very worth studying.

Some fascinating material is contained in a rather posh booklet on speakers published by a newcomer to the U.S. audio scene, the French company known simply as 3A. Its U.S. address is 172 Madison Ave., New York, N.Y. 10016.

A big catalogue bulging with tape, audio, record and guitar accessories has been issued by Le-Bo Products, Co., Inc., Maspeth, N.Y. 11378. It is impossible to describe all the goodies shown in this booklet—they include things from blue-denim carrying cases to splicing tape.

Switchcraft, Inc., 5555 No. Elston Ave., Chicago, Illinois 60630, has released a new catalogue, no. C-510, which describes a number of its connectors and includes a useful mating chart showing what mates with what. This company also has a one sheet release describing its own new stereo selector system, the model 640R, which is designed for handling up to 1728 combinations of stereo sources, amplifiers/receivers and speaker systems.

JVC has a paper that goes deeply into the biphonic type of sound which is an extension of binaural sound (described here in the June 1977 issue). If you're interested, contact JVC America, Inc., 58-75 Queens Midtown Expressway, Maspeth, N.Y. 11378.

SAE's full-line brochure describes it equalizers, tuner, noise-reducer, electronic crossover, etc., in addition to preamps and power amps. SAE's full name is Scientific Audio Electronics, Inc. and they are at P.O. Box 60271, Terminal Annex, Los Angeles, Calif. 90060.

An increasing number of British audio companies have indicated interest in the U.S. audio market. For information on them, write to the British Information Services, 845 Third Avenue, New York, N.Y. 10022.

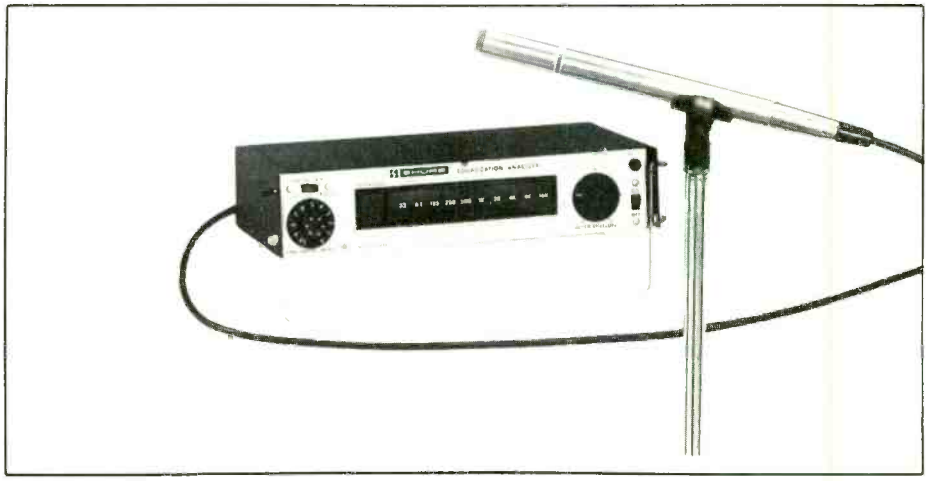
Finally, there's the latest edition of that perennial electronic "dream book," the Heathkit catalogue. Write for a copy (Catalog 816) to Heath Co., Benton Harbor, Michigan 49022.

MUSICAL NEWSICALS

The primary trade show of the National Association of Music Merchants was held this June 10 through 12 in Atlanta rather than in Chicago as usual. The Atlanta show included a considerable number of first-time exhibitors, although many of them were smaller companies based in the South who may not have exhibited at previous Chicago NAMM shows due to the high transportation costs involved, in addition to newly-formed companies. This year's show followed the pattern of the last several years in its increased emphasis on PA equipment. So much interesting new PA gear was shown, in fact, that it will take the better part of two columns to catch up on it all. This month, however, we'll begin by looking at the new musical instruments and synthesizer products shown in Atlanta, and we'll continue on into musical instrument amplifiers and accessories and PA equipment in the next two months' columns.

– SOUND REINFORCEMENT –

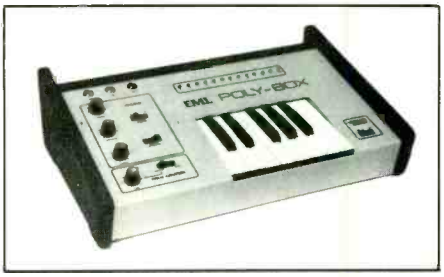
New from Shure Brothers Inc. (222 Hartrey Avenue, Evanston, Ill. 60204) are a pair of units which combine to form a complete room analysis and equalization system. First is the Model SR107 equalizer ($250.00). This unit is a solid-state, active equalizer, whose ten octave-band filters have center frequencies spaced at octave intervals from 31 Hz to 16 kHz. Each of the ten filters has a ±15 dB boost or cut range at its center frequency, and was designed for minimum phase characteristics. Other features include a master level control which varies the level of the equalized output over a ±15 dB range, an LED peak overload indicator and an equalizer bypass switch. A low impedance, line-level input and a low impedance mic/line output are accessed via XLR connectors on the back panel of the unit, and three-conductor ¼-inch phone jacks are provided for high impedance line in and line out. The SR 107 mounts in a 19-inch rack, and takes up only 1-3/4 inches of vertical rack space. The other part of Shure's room equalization system is the M615AS Equalization Analyzer System ($715.00) which consists of the ES615 omni-directional Analyzer Microphone and the M615 Equalization Analyzer (both available individually). Shure has taken an original approach to displaying data about the sound system's frequency response; rather than using the usual array of five or more indicator lamps per equalization band to display the actual signal levels, the M615 uses only two LEDs per band to indicate whether the system's response falls above or below a variable, pre-set "Hi/Lo Envelope." To use the M615 system, the analyzer's built-in pink-noise generator is connected to an input of the sound system and the ES615 or other calibrated mic is connected to the analyzer's input. The user then adjusts the equalizer's controls to extinguish the LEDs for each frequency band, indicating that the response in that band falls within the "Hi/Lo Envelope" width which has been preset. This "envelope," which is best thought of as a "window" or a "dead band," is continuously variable from 12 dB to only 2 dB in width. By starting with the "window" at its widest and progressively narrowing it while adjusting the equalizer as necessary to keep all the LEDs turned off, it is possible to equalize a sound system's octave-band pink-noise response to within ±1 dB in a matter of minutes.



CIRCLE 18 ON READER SERVICE CARD

– ACCESSORIES –

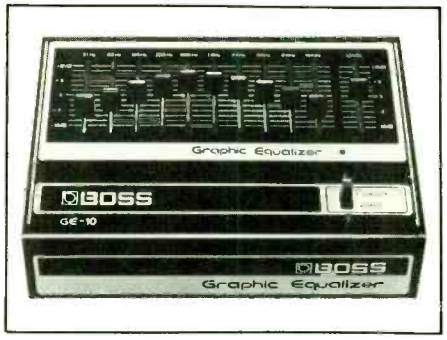
Electronic Music Labs (P.O. Box H, Vernon, Ct. 06066), who make the Syn-Key synthesizers, have introduced the EML Poly-Box ($475.00) which is a device designed to give polyphonic capabilities to monophonic synthesizers. The Poly-Box is a digital/analog device which is programmed via a one-octave



keyboard to produce harmony notes or chords whenever a single note is played on the main synthesizer keyboard. The harmonic intervals of these additional notes are set via the Poly-Box keyboard, with a range of three octaves above to one octave below the original note. The Poly-Box's output precisely follows the main synthesizer's oscillator through portamento, vibrato and keyboard transpositions. Manual tuning, variable phasing and a low-pass filter are included in the Poly-Box to give maximum control and wide variability of the sounds produced. The unit interfaces directly with patchable synthesizers, and smaller, non-patchable instruments may be simply modified to interface.

CIRCLE 8 ON READER SERVICE CARD

A new graphic equalizer, the Boss GE-10, has been introduced by Beckman Musical Instruments (P.O. Box 22289, East Los Angeles, Ca. 90022). The unit is an octave band model and has a ±12 dB cut and boost range for each band.



Each of the ten sliders has a detent in the center "flat" position, and there is a bypass switch for ease and convenience in setting up and adjusting the equalization. An input level switch is provided to match the GE-10's sensitivity with microphone, musical instrument or line level inputs for maximum versatility. The GE-10 is particularly suited for use with musical instruments because it has a remote switch jack for the bypass switch function, allowing the use of the unit like a special effect with a separate footswitch.

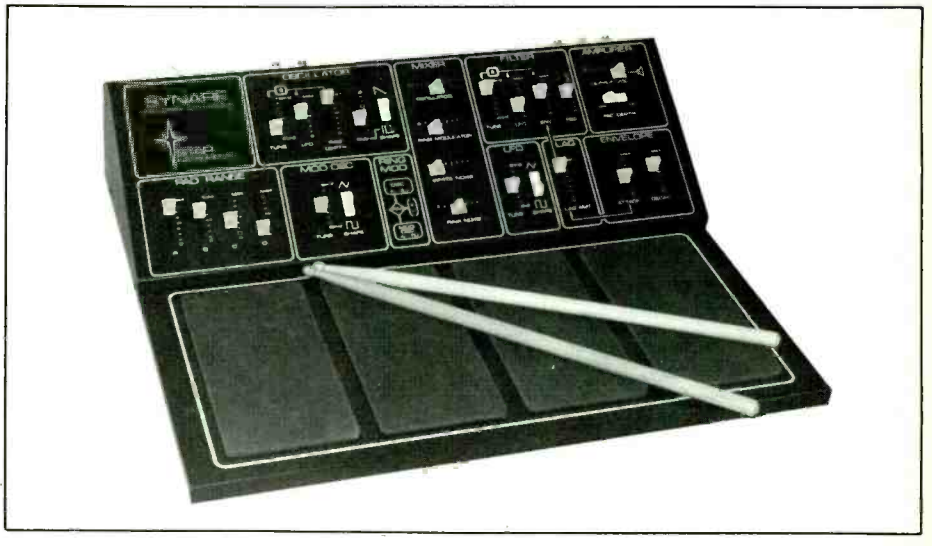
CIRCLE 14 ON READER SERVICE CARD

TEAC Corporation of America (7733 Telegraph Rd., Montebello, Ca. 90640) has announced the introduction of an expander unit for use with the Tascam Model 5 mixer. The Model 5EX expander ($1300) contains eight Model 201 input modules to double the input capacity from eight to sixteen; an additional four input modules may be added as an option, bringing the total to twenty

inputs. The 5EX is self-powered, but connects into the Model 5's four output buses and cue, echo and solo buses.

CIRCLE 19 ON READER SERVICE CARD

Star Instruments, Inc. (P.O. Box 71, Stafford Springs, Ct. 06076) has an interesting product in their "Synare" Percussion Synthesizer ($795.00). Four rubber pads are provided on the front of the unit to trigger the synthesizer; each of the four pads is individually tunable. In addition, each pad has three depth "zones" which can be set individually so that different sounds are produced from a combination of a triangle/square /pulse voltage-controlled oscillator, white noise generator and pink noise generator. The VCO signal can be processed with a ring modulator with its own triangle/ square-pulse oscillator, and the VCO-plus-noise signal is processed through



a VCF with variable resonance and a dual VCA. The 24 dB/octave VCF is controlled by any combination of manual control, triangle/square LFO, and envelope generator with lag, and the VCF becomes a sine-wave oscillator at the maximum resonance setting. The entire unit is packaged in a compact eighteen-pound housing which is

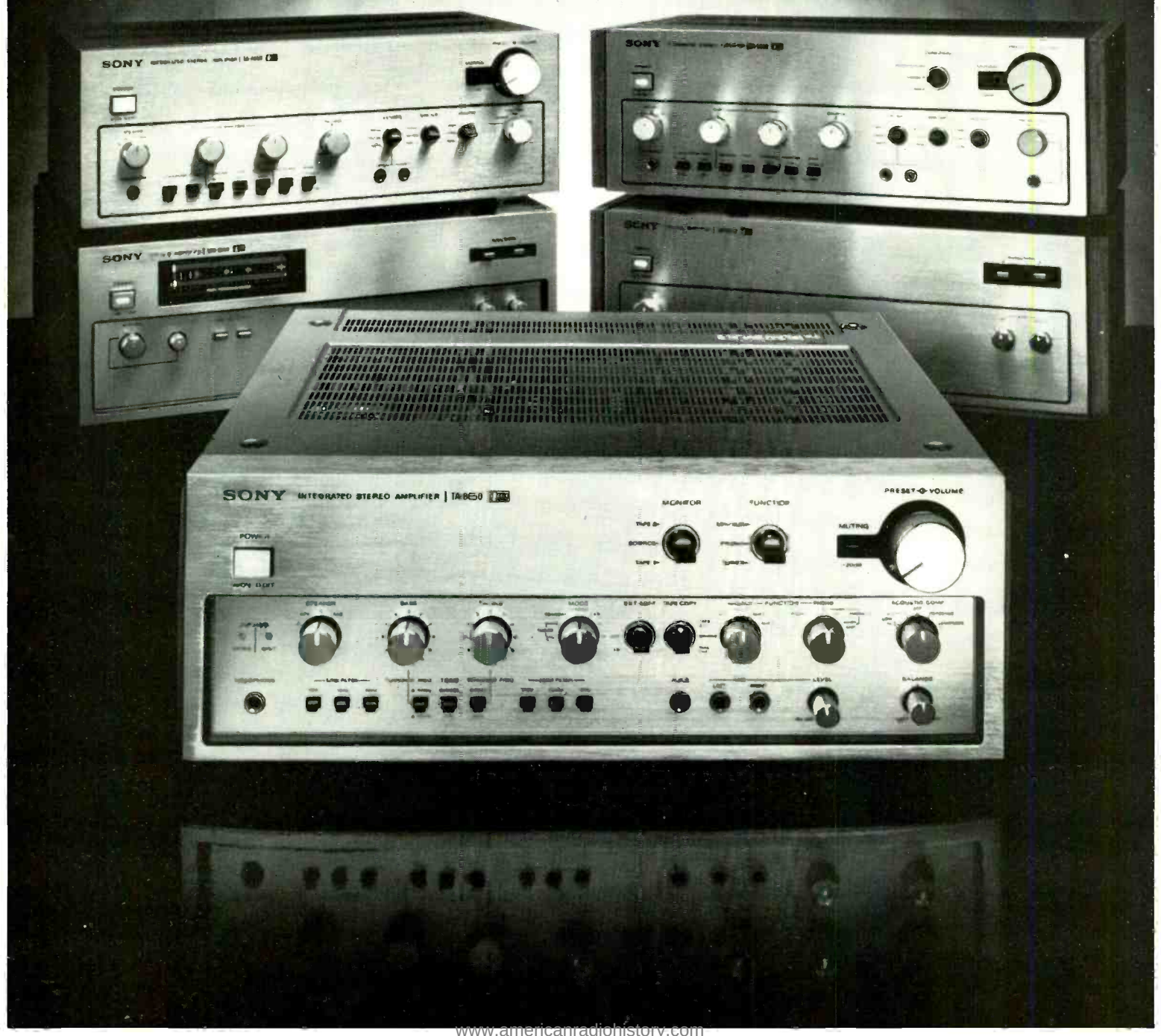
CIRCLE 9 ON READER SERVICE CARD

designed to mount on a tom-tom stand.

A lesser-known name in road cases is Bobadilla Cases (2302 East 38th Street, Vernon, Ca. 90058) who offers a full line of foam-lined fiber cases in addition to the plastic-coated plywood models for the budget-minded. The Bobadilla catalog lists an extensive selection of standard-size cases in their standard construction, which includes ABS-clad plywood in one of six colors, polyester foam lining, riveted construction and heavy-duty recessed hardware. Bobadilla's specialty, however, is their extensive custom-design program. Even for standard-size cases, custom design offers the ability to specify exactly the features and construction you want and need, ranging from extra-thick foam with crushed velvet lining in a guitar case to piano hinges and heavy-duty corners on an amplifier case. One construction detail unique to Bobadilla cases is their "Double Edge" corners. Most other road cases use butted corners which are actually held together by rivets driven through the metal edging strip and plywood at the corners. Double Edge cases use an extruded aluminum channel with an edge on the inside as well as the outside of the case; the edges of the ABS-clad plywood are mitered at a 45-degree angle to fit into the groove in the edging channel, and the case is constructed by gluing the panels into the edging. This type of construction is said to result in a stiffer, stronger case which is almost totally watertight if the optional watertight valence is ordered. As an additional service, Bobadilla maintains a file of dimensions for all custom cases they build so that it is a simple matter to replace or duplicate a custom design at a later date.

CIRCLE 4 ON READER SERVICE CARD

V-Fets are the new generation of transistors.



And you know how hard it is to understand the new generation.

Remember the phrase "generation gap"? Well it's not only true for generations of men, but generations of machines, too.

V-Fet devices are a major advancement, needing major explanation. And nobody is more equipped to offer it, than Sony.

Sony pioneered the first generation of transistors, some 25 years ago.

Today, Sony is predictably innovative again, being: the first to offer V-Fet equipment commercially. And the only ones to bedazzle you with a whole line of it.

So with these credentials behind us, we will begin our explanation of the new generation.

First came the Fets.

The new generation really began many generations ago. Fets—or field effect transistors—were first conceived in the 1920's. But the concept was so far ahead of its time that nobody quite knew how to execute it.

Fet's work quite differently than the bipolar transistor; the transistor you're familiar with. The bipolar transistor works by conducting a small amount of current, which then induces a high level of current. With the Fet a small amount of voltage (rather than current) controls the high level of current.

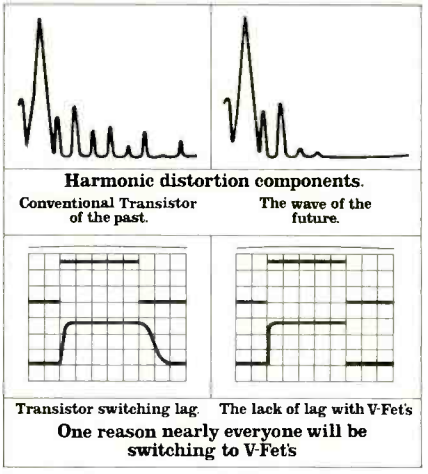
This bestows a Fet with high speed reaction time. Regular transistors have a delay in reaction time, creating problems like notch distortion and TIM (transient intermodulation) distortion.

This high speed reaction means also that Fets are extremely efficient and accurate in the high frequency range. Therefore they allow more precise and stable negative feedback, and minimal distortion.

And, to heat up the argument, a Fet will never be afflicted by thermal runaway. High temperature does not induce the self-destructive current surge that you'll find in the regular transistor.

V-Fets. Or, bye, bye, bipolar.

Wondering why Fets have not taken over, with the transistor becoming a part of history?



Harmonic distortion components.

Conventional Transistor of the past. The wave of the future.

Transistor switching lag. The lack of lag with V-Fet's

One reason nearly everyone will be switching to V-Fet's

Well, for one reason, a Fet will not allow high currents to pass through it. And today's loudspeakers demand high currents to drive them.

Enter the V-Fet. Vertical field effect transistor. In this structure, thousands of Fets are ordered in a parallel orientation. The current passes through the silicone chips *vertically.*

Thus, the ability of the V-Fet to handle a lot of current is many times greater than that of small signal Fets—like the kind found in FM tuners and pre-amps.

Sony made it possible for this complex network to be mass produced, by devising the "Selective Oxidation Process." A new technology originally developed for manufacturing large scale integrated circuits.

Sony's V-Fets. A full line, from A to V.

Sony makes both integrated amps and power amps with V-Fet circuitry. The TA-4650, TA-5650, TA-8650, TAN-5550 and TAN-8550.

But if you rest your purchase decision on specs alone, V-Fets will disappoint you.

For example, Sony makes two amps, one V-Fet, one not, with identical specs. Yet the V-Fet amp costs more than a spec more.

Obviously, the true measure of V-Fets can't be measured by anything except the human ear.

Now that you've listened to us, really listen to us.

So go ahead and measure it! Bring your favorite record to your V-Fet dealer. Ask him to play it. You'll find your favorite record will become even more of a favorite, as the sound opens up to you like never before.

And, if you want to open up a brochure on V-Fets, we'll send you one. Write to SONY, 9 West 57th Street, New York, New York 10019.

We have one note to add. V-Fet equipment is not cheap. So if you've appreciated our explanation, you'll find that a little knowledge can be an expensive thing.

SONY®
V-FETS

A Psychoacoustic Phenomenon the Aphex

AURAL EXCITER

By Michael Gershman

In 1956, Curt Knoppel incorrectly wired up a sound amplifier in kit form; he eventually corrected his mistake, but not before he had heard an intriguing sound, the sensation of which stayed with him for twenty years.

Over the course of those two decades, Knoppel periodically set aside inventing time (he participated in the design of the nose cone for the Discoverer satellite and various space hardware) and spent nearly $100,000 of his own money chasing that elusive thread of sound wherever it might lead.

In 1971, Knoppel read an account of the discovery of holography and had a galvanizing brainstorm: The tantalizing distorted sound he had heard years before might be to pure sound what holography was to pure light.

His basic theory was proven correct, and subsequent research led him into the realm of psychoacoustics—the field of study that concerns itself with how the brain processes sound. Now he had the final parts of his equation: The human brain was obviously an integral element in "hearing" sounds that could not be measured on oscilloscopes.

Based on this insight and his persistence, what is now the Aphex Aural Exciter was born in the spring of 1973.

Because of the Aphex Aural Exciter, people who buy the records of such artists as Linda Ronstadt, James Taylor, Jackson Browne and Donald Byrd, or attend the concerts of Paul McCartney & Wings, Supertramp, Kiss, Crosby, Stills & Nash and others, hear "enhanced" sound.

Record producers, engineers and recording artists use terms like "brightness," "warmth," "increased presence" and "fullness" to describe what Aphex does to sound.

Aphex—an acronym for Aural Perception Heterodyne Exciter—addresses psychoacoustic circuitry in the listener. The device gives the brain credit for what it has always done—constantly emphasize and de-empha-

size fragile bits of acoustic information to create individual realities for each listener. Aphex further stimulates psychoacoustic activity to enhance perception of acoustic material.

This is a revolutionary development in state-of-the-art audio. Psychoacoustics is a little-discussed (and less understood) field. Using a psychoacoustic device on commercial records is indeed a major step in the history of sound recording.

Aphex is used in music and speech applications for entertainment in combination with the mixing facility. For professional use, it is employed in a fold-back configuration like an echo unit. The instrument has standard balanced inputs and outputs and is designed for a level of 4 dBm=0 VU. It is suitable for stereo mixing; two units would be required for a quad mix.

During the mixdown, Aphex is folded back onto the main channel, at a level 20 dB lower, taking on the character of a sub-carrier. A differential input is thereby created and, upon processing by the listener's aural circuitry, creates interestingly unique and "beneficial" effects.

To achieve the optimum effect, engineers set each mix component at the desired levels and keep the Aphex mix and main return in set positions; thus, by moving the send signals up and down, the engineer controls the amount of Aphex on each track. Aphex can also be used as an "instrument" on a separate track.

There is also an application for Aphex in the mastering phase. Working with already mixed material, the mastering engineer can use Aphex in "blanket" fashion to give some punch to lifeless tracks. However, since Aphex is transient sensitive, it is quickly noticeable on percussion tracks, especially cymbals, limiting its use in mastering.

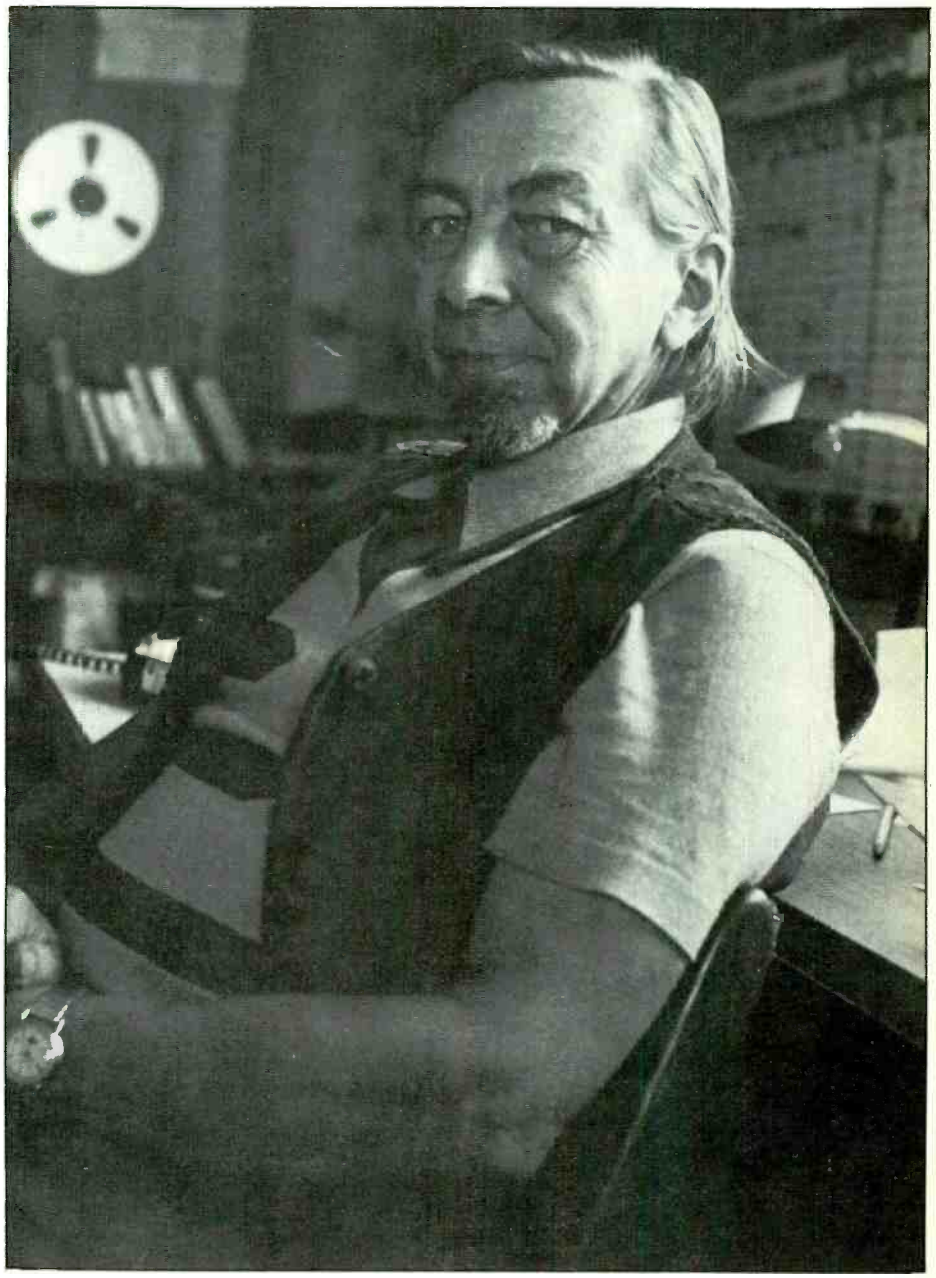
There are other drawbacks to Aphex. It will make noisy tracks even noisier; if an instrument is slightly out of tune, Aphex will make it more objectionable; it can sound harsh when applied without subtlety. Says Knoppel, "Aphex is a very dumb instrument. It simply doesn't know that noise isn't beautiful."

What Knoppel says on the plus side is, "With 30 ips tape that is clean, free of unwanted distortion and has an acceptable signal-to-noise ratio, Aphex will tend to make the sound considerably more acoustical—like the original sound source. Pieces of acoustic information that get lost in processing audio are resurrected and emulate their "live" ambience. Aphex doesn't create information. What it does is seek out information and make it audible. Then, the miraculous circuitry in the brain further enhances the information to give vocals, percussive instruments and acoustic instruments greater realism."

The invention of Aphex is due to a fantastic combination of coincidences which occurred over a twenty-year period. In 1956, Knoppel was wiring up a stereo amplifier kit. He didn't notice that one of the coupling bridges went from left to right and the other from right to left. Consequently, a lot of the cables went to the wrong points and resulted in low plate voltage. (The unit was designed for 200 volts and instead got only six volts; as a result, certain tubes were starved for plate voltage.)

Coincidentally, in trying to make the amp work, he used a stereo phono cartridge as a sound source. The coil of the cartridge supplied the inductance needed under those conditions. The mis-wiring and cartridge added up to a fantastic coincidence leading to a revolutionary sound improvement.

Knoppel remembers the moment well. "I will never ever forget the potency of that sound. It was almost something not of this world. I was



Photos courtesy of Aphex, Ltd.

Aphex creator Curt Knoppel: ". . . I knew I was on the right track."

completely fascinated with it and kept on playing with it. I tried to figure out what was what, but eventually I just put it away."

A native of Stockholm, Sweden, Knoppel moved to America two years later and was still intrigued enough to bring the strange amplifier kit along . . . just in case.

Fully twelve years later, after a series of triumphant inventions—a plastic automobile muffler, the Discoverer satellite nose cone and NASA space suits—Knoppel decided to investigate the mystery again.

"I took it down from the attic one night and tried to recreate that incredible sound as well as I could. It didn't work until I tried a phono source again. At the time, I didn't realize how important it was."

Although he was personally stumped, his inventions had created enough capital for him to hire his own acoustic experts. From the spring of 1968 until October 1970, Knoppel poured some $85,000 into the project. The acoustical engineers he hired could only shake their heads in bewilderment. Says Knoppel, "I was forced to accept the conclusion that the sound was created by voodoo and abandoned the project."

An account of Dr. Dennis Gabor's discovery of the principles of holography convinced him to abandon the abandonment.

"I read the account in a newspaper and a light went on in my head. It made me think that my accident could lead to a discovery in sound equivalent to what holography meant to light."

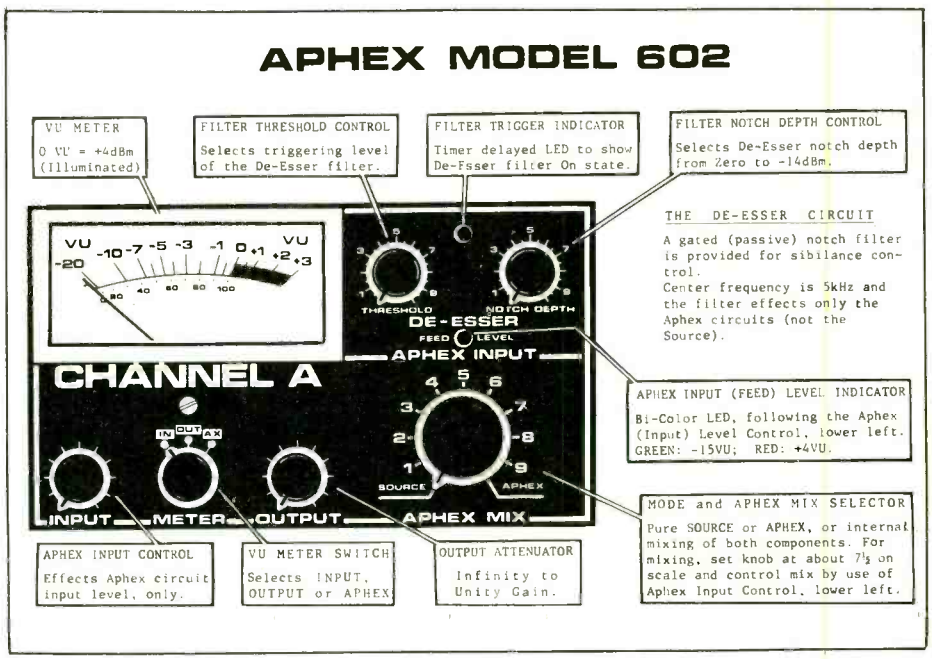
"Then I got the notion that there was still something missing in my equation—the human ear. It got me looking beyond electronics for the first time and I started reading about the hearing process. When I discovered the science of psychoacoustics, I knew I was on the right track."

From that point, he had the basic direction to the invention of Aphex, but there were still circuitry kinks to be worked out. He filed the first original patent in the spring of 1972, but three more years of work were required before a reliable working model was available for demonstration in April, 1975.

The Aphex 402, an earlier version of the current model 602.

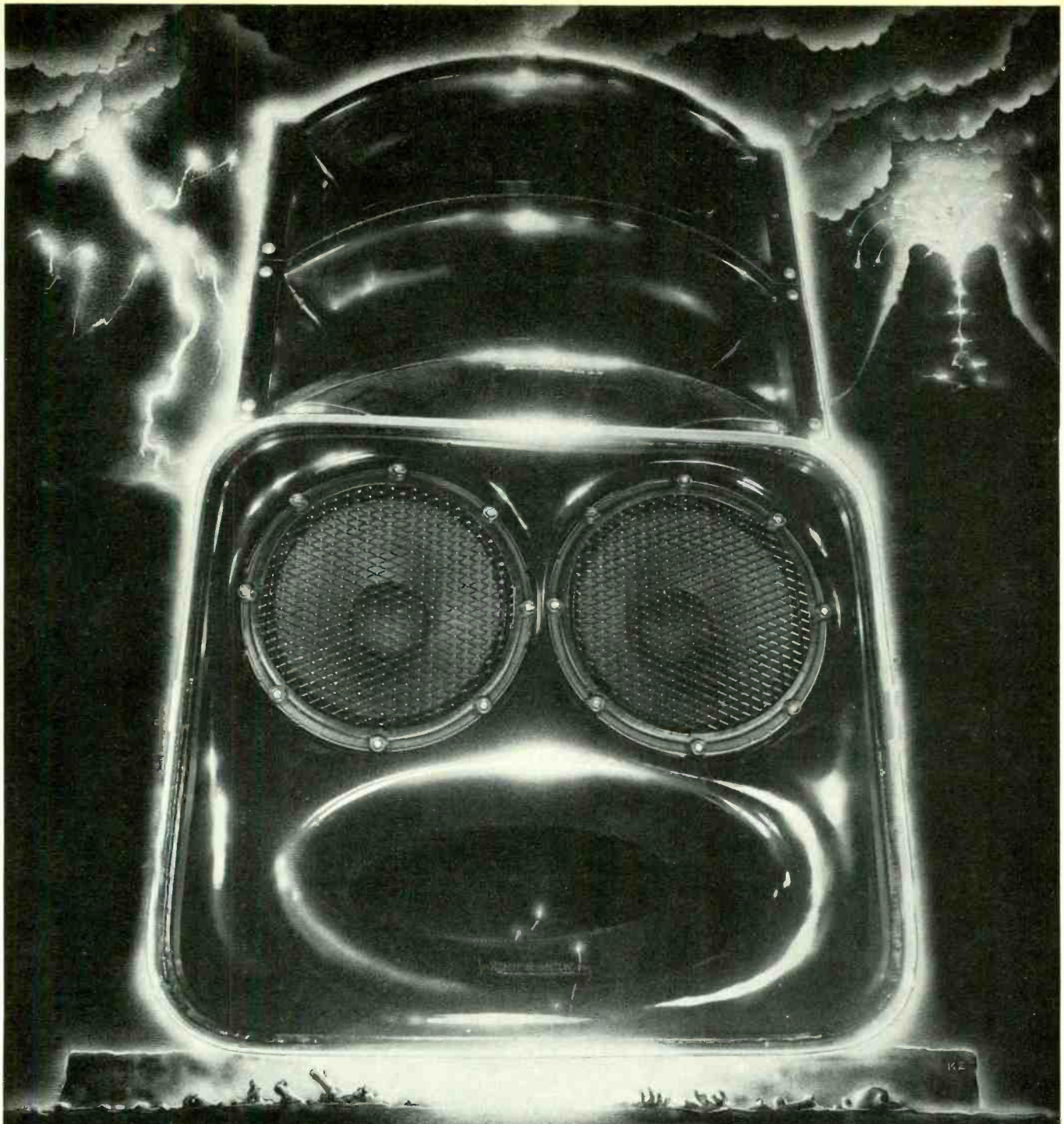
connection didn't work out, but Knoppel demonstrated Aphex for Caesar with gratifying results. The 24-year-old Caesar had both the musical interest and the business experience to fully appreciate the possibilities; his infusion of capital and energy gave the project a second wind.

The two formed Aphex Systems, Ltd. in March 1975 in Framingham, Mass. In less than a year, Aphex got its first break when the system was used on NBC-TV's Midnight Special. (It has been used on every broadcast since then.)



Once again, a coincidence helped Aphex along at a crucial point. Knoppel was introduced to Marvin Caesar, an accountant and land developer on another business matter. The original

That April, Aphex was used by Showco, Inc. a leading sound company, for "live" performances by Paul McCartney & Wings. Record producer Peter Asher heard the Wings system



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CIRCLE 95 ON READER SERVICE CARD

SPECIFICATIONS

2 CHANNELS, 3½x19" RACK MOUNT

INPUT CONDITIONS
(Internal level boost switch on PC Card 620)
NORMAL LEVEL: 0 VU= +4dBm; BOOSTED: 0 VU= -8dBm.
IMPEDANCE: 20kOhms nominal. HEADROOM: 19dB.
Input control effects Aphex feed only.
CONNECTORS: XLR, pin 1 gnd, pin 2 low, pin 3 high.

INPUT INDICATOR
Bi-color LED in Aphex feed circuit.
GREEN: -15 VU; RED: +4 VU.

SOURCE SIGNAL CONDITIONS
ACTIVE BYPASS: (Mix Control at "Source") unity gain.
INTERNAL MIX: (Mix Control at 7½) 6dB loss.
SIGNAL to NOISE LEVEL: -95dBm

OUTPUT CONDITIONS
IMPEDANCE: 600 Ohms, transformer, floating.
CONTROL: Fully Clockwise (10) unity gain.
PHASE: Rear panel phase reversal switch.
CONNECTORS: XLR, pin 1 gnd, pin 2 low, pin 3 high.

DE-ESSER CONDITIONS
Gated Notch Filter for sibilance control.
CENTER FREQUENCY: 5 kHz
NOTCH WIDTH: 1 kHz to 20 kHz
NOTCH DEPTH: (Adjustable) 0 to -14dB
THRESHOLD: (Adjustable) full range.
FUNCTION INDICATOR: LED, timer effected.
RISE TIME: 2 Microseconds.
TRIGGERING: Via 5kHz bandpass filter.
Signal path through De-esser is passive and effects Aphex feed, only.

VU METER CONDITIONS
Meter circuits are buffered.
CONTROL: Input/Output/Aphex Circuit Output
Aphex position padded 7dB (meter protection)

POWER CONDITIONS
115V / 230V AC, 25 Watt.
Rear mounted power ground lifter switch.

and decided to use Aphex on his then upcoming album with Linda Ronstadt, *Hasten Down the Wind.* Aphex was also used on the floor of the Democratic National Convention at the recommendation of Republican Senator Lowell Weicker.

Since then, Aphex has been used by a full range of musical performers—MOR artists like Neil Sedaka and Johnny Mathis, rockers like Led Zeppelin and The Grateful Dead and country artists like Dolly Parton and Hank Snow.

Record producers and engineers have been quick to recognize the utility of Aphex. Producer/engineer Val Garay, who has used Aphex on projects for Ronstadt, Eric Carmen, Andrew Gold and James Taylor, remembers his introduction to the process.

"Peter Asher was producing *Hasten Down the Wind* and we had already mixed four of the cuts when he heard Aphex used on the Wings tour. He called me up and told me we had to use the unit. We went to the Mastering Lab and started working with it on already mixed material. Joni Mitchell came running in when she heard it. Linda went crazy for it and we used it on the rest of her album."

"In my business, the ideal of course is 100% perfect sound. When you look at it that way, a two or three per cent improvement is just incredible. When someone hands you three per cent in a little black box called Aphex, it's like a gift from God."

Other sound technicians agree, and Aphex is now a permanent fixture at studios like Wally Heider's, The Record Plant and A&M Records. Similarly, it is used "live" for all performances by America, Glen Campbell, Kiss, Crosby, Stills & Nash, etc.

The specialized unit is even marketed in a special way. Says Caesar, "We don't rent the hardware, but rather we lease it on the basis of $30 per minute for finished product. We charge a licensing fee to reproduce our process and our effect. For the average album, it works out to be about $1,000."

Producers, engineers and musicians all agree that $1,000 spent on Aphex is $1,000 well spent. Rod Stewart heard the Aural Exciter demonstrated and told his team of sound men, "Aphex is a must."

What Every Studio Should Have In Common.

These studios are owned by businessmen who have at least one thing in common...

an unrelenting commitment to protect and build on their investment. In these times of growing competition and increased customer demands, success only comes with hard work and a uniquely developed talent.

For over two years THE EXPRESS SOUND CO. has specialized in the service and support of that talent. From the beginning, ours has been a focused effort to create and customize the kind of company these studio owners (and others like them) look to for their equipment, consultation and service needs.

If the time has come for you to join this special community of professionals, you'll have the opportunity to share in the experiences and talents common to them all... technical knowhow, seemingly endless hours applying it and, most important, profitably selling and delivering it.

AND THERE IS ONE MORE THING EVERY STUDIO SHOULD HAVE IN COMMON . . .

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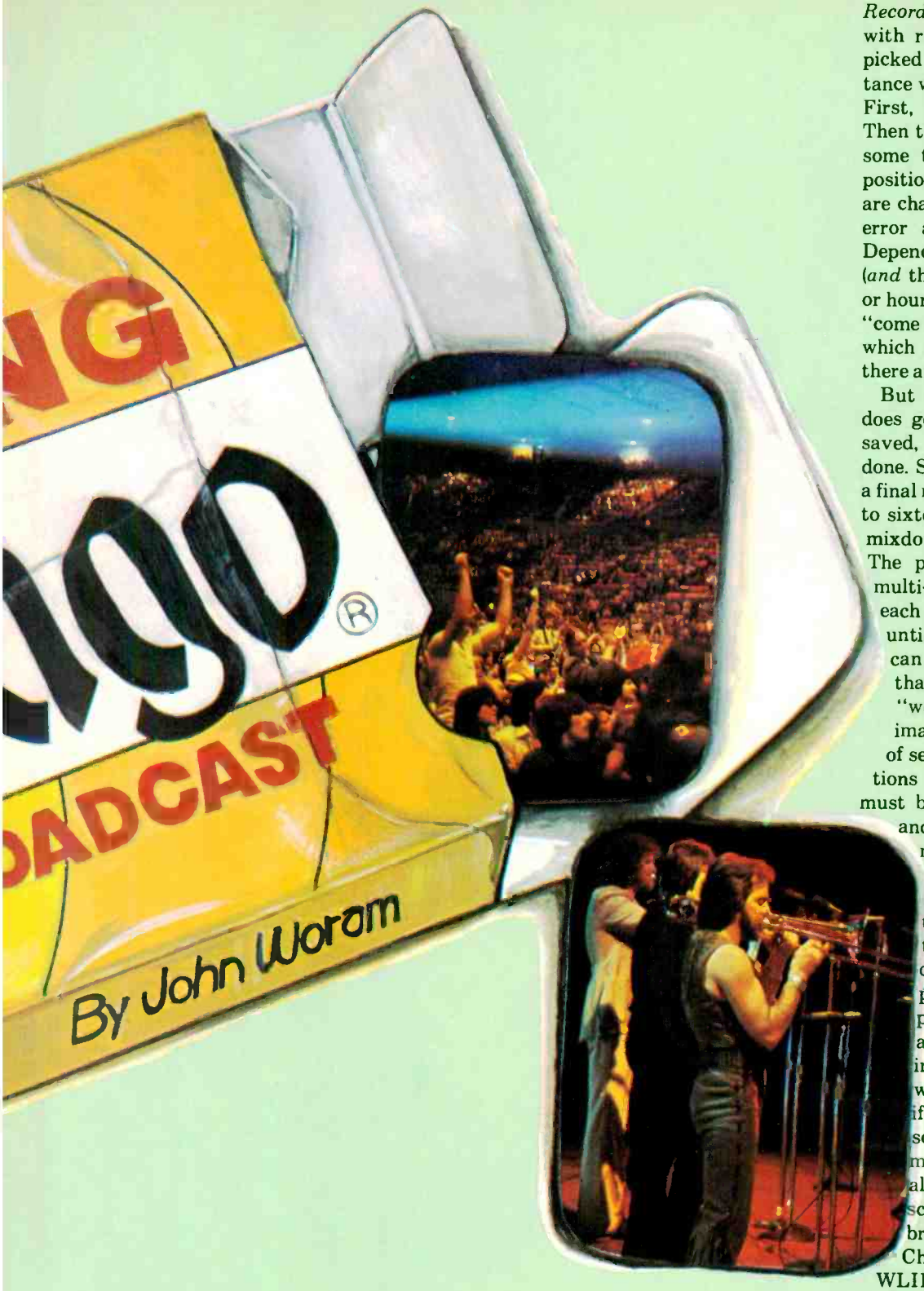
CIRCLE 83 ON READER SERVICE CARD



Many—perhaps even most—*Modern Recording* readers have been dabbling with recording long enough to have picked up at least a nodding acquaintance with general session procedures. First, you set up the microphones. Then the musicians gather round, and some time is spent experimenting—positions are shuffled, microphones are changed, and after some trial-and-error a basic "sound" is achieved. Depending on the skills of the engineer (*and* the group) this can take minutes or hours. Once in awhile it just doesn't "come together" for quite a long time, which can be a bit embarrassing if there are a few visitors present.

But eventually a series of "takes" does get recorded. The best ones are saved, and perhaps some editing is done. Sooner or later (often, it's later), a final master with anywhere from four to sixteen or more tracks is ready for mixdown.

The production advantages of this multi-track technique are obvious—each part can be done, and re-done, until it's perfect. And, the engineer can spend hours later on honoring that famous procedure known as "we'll fix it in the mix." But now, imagine a somewhat different kind of session, with the following conditions imposed: (1) All microphones must be set up properly in advance, and may not be moved once the musicians show up. (2) There will be no rehearsal time. (3) All mics will be routed to a two-track stereo output, and the entire "balancing act" must be done while the musicians are playing. (4) Re-takes are not permitted. (5) Over a quarter-of-a-million "visitors" will be judging your work. (6) The group is world famous, and so will you be if you screw things up. If this sounds like an engineer's nightmare, you may be right. But it's also a reasonably accurate description of the "live" radio broadcast featuring the group Chicago, over radio station WLIR-FM, on Friday, May 20, 1977.

The occasion is a benefit concert for the One-to-One Foundation, an organization whose purpose is to help keep retarded and emotionally disturbed children out of places like the now-notorious Willowbrook institute. So far, One-to-One's biggest fund-raising events have been concerts and mini-telethons, and this Coliseum show eventually raised $145,000 to aid them in their work.

The "live" Chicago radio concert was one of many special broadcasts done in conjunction with WLIR's regular weekly radio concert series—the longest running show of its kind in the country having started back in August of 1971 with weekly "live" broadcasts from the Ultra-Sonic Recording Studios in Hempstead, New York. At that time, the audience would gather in the studio, along with the musicians. But as the series expanded, it became harder to get groups who could find the time to leave their tours and enter the studio for a radio broadcast. The situation necessitated going "on location" to broadcast the groups from the concert facilities in the Long Island, N.Y. area.

Engineering the broadcast is Michael Colchamiro, better known as Michael "Tapes" for the benefit of those who can't handle Colchamiro. Michael has been producing and engineering WLIR's concert series since its start when he was a staff engineer for Ultra-Sonic Recording. When the series left Ultra-Sonic 3½

Console Input Assignments

Console Input	Instrument	Pan Position
1	Bass drum	Center
2	Snare drum	Left center
3	Small rack tom	Left
4	Medium rack tom	Left/left center
5	Large rack tom 1	Left center
6	Large rack tom 2	Center/Right center
7	Floor tom	Right center
8	Roto-tom	Right
9	High hat	Right center
10	Over-all drums 1	Left center
11	Over-all drums 2	Right center
12	Bass direct	Center
13	Instrumental sub-mix	Right center
14	Vocal sub-mix	Center
15	Horn sub-mix	Left center
16	Percussion sub-mix	Left
17	Full P.A. mix	Not used
18	Stage left mic	Left
19	Stage right mic	Right
20	Audience left	Left center
21	Audience right	Right center

years ago, Michael and Audio by Zimet (a retail high-end audio shop) put together the Zimet Mobile Recording Van to be used exclusively for the production of radio concerts.

Getting It Right

The Zimet van is not quite the massive tractor-trailer affair, over-

weight because of a super board and twin 24-trackers. Remember, these are "live" radio concerts, and although "we'll fix it in the mix" still applies, the mix is not going to be done three weeks from tomorrow. It's going to be done *now*. The ground rules are ominously simple: Get it right the first time, or don't get it at all. This rules out the need (and the excuse) for multitrack, and forces everyone to regard the concert in musical terms and not as an exercise in data processing.

Therefore, a reliable console and some assorted odds and ends are about all that's required. At the moment, a modified Tascam Model 10 console with expander is on board. In-house modifications to the 24-input board include a solo system, which is especially vital for "live" work.

The monitor system is comprised of Dynaco amps, and Dynaco A-35 speakers modified with an additional piezo-electric tweeter and custom-built crossovers. In a somewhat unorthodox departure from conventional control room monitoring, Michael supplements the speaker monitors with a pair of Sennheiser 414 "open-air" headphones, whose foam air-cushions allow him to hear the speakers even while wearing the phones. He finds the phones help him pick out the details, and with several years worth of shows behind him, he's learned to equate this phones-plus-speaker sound with what he would hear under more traditional circumstances.

Peripheral equipment installed in the van includes a Sound Workshop reverberation system and also delay lines, dbx 160 series compressors, Soundcraftsmen 2212 graphic equalizers and custom-built distribution amplifiers. A pair of Shure M63 Audio Master control centers act as the final drive stage before the phone lines.

Getting It Through

And speaking of phone lines, they have been installed a few days in advance. As you may suspect, its not quite enough to tap into the nearest pay phone, as these lines have quite a limited response. The special lines are good up to about 15 kHz, and the first task of the day is to check them for phase, frequency response, signal-to-noise and headroom—all of which may vary depending on the conditions of the lines, and perhaps the mood of the phone company. An additional regular line is set up to the radio station for instant communication during the concert. (If disaster strikes, one doesn't want to have to get on line at that pay phone in the lobby.)

Miking For Broadcast

For a one-shot "live" broadcast, microphone placement technique varies considerably from the usual studio-type session. Michael had never before worked with Chicago, but fortunately, Chicago *had* been working with Clair Brothers Audio [the sound-reinforcement company] consistently. Through discussions with Gene Clair, it was decided not to deviate from Chicago's normal stage miking.

So, the microphone set-up was a two-part mixture of educated guess-work and feeds from the Clair Brothers' P.A. set-up. The sound company had placed eleven microphones on the drums, and these were fed through custom-built mic/line split boxes so that both the console in the Zimet van and Clair Brothers' P.A. board could derive separate drum mixes. The P.A. mix would be mono, while the broadcast mix would be stereo. A direct pickup from the electric bass was also split to both boards.

To complete the pickup, Clair Brothers supplied sub-mixes of horns, percussion, vocals and "everything else." Each of these was routed through their board first, and then out to the van. Therefore, it was up to the sound company to feed a useable set of sub-mixes out to the van, and for Michael to blend them together into a satisfactory broadcast feed. Additional microphones for broadcast-only use were set up on stage left and stage right, and two microphones were placed to cap-

ture audience reaction. And as a final hedge against everything that can go wrong once you're on the air, a feed of the entire P.A. mix was also routed out to the van for emergency use. Fortunately, it wasn't needed.

The basic drum mix in the van was provisionally established with the help of a volunteer drummer, who made the rounds of the drums before the group arrived. All other mixing was done *after* the actual concert began.

When trying to create a good mix, it helps to know the group's music, and although Michael is quite familiar with Chicago, he still spent a few pre-session hours reviewing their recent recordings. It also helps to be working with a top-notch sound company such as Clair Brothers, since their work can spell the difference between an adequate (or worse) pickup and an excellent one.

On the Air

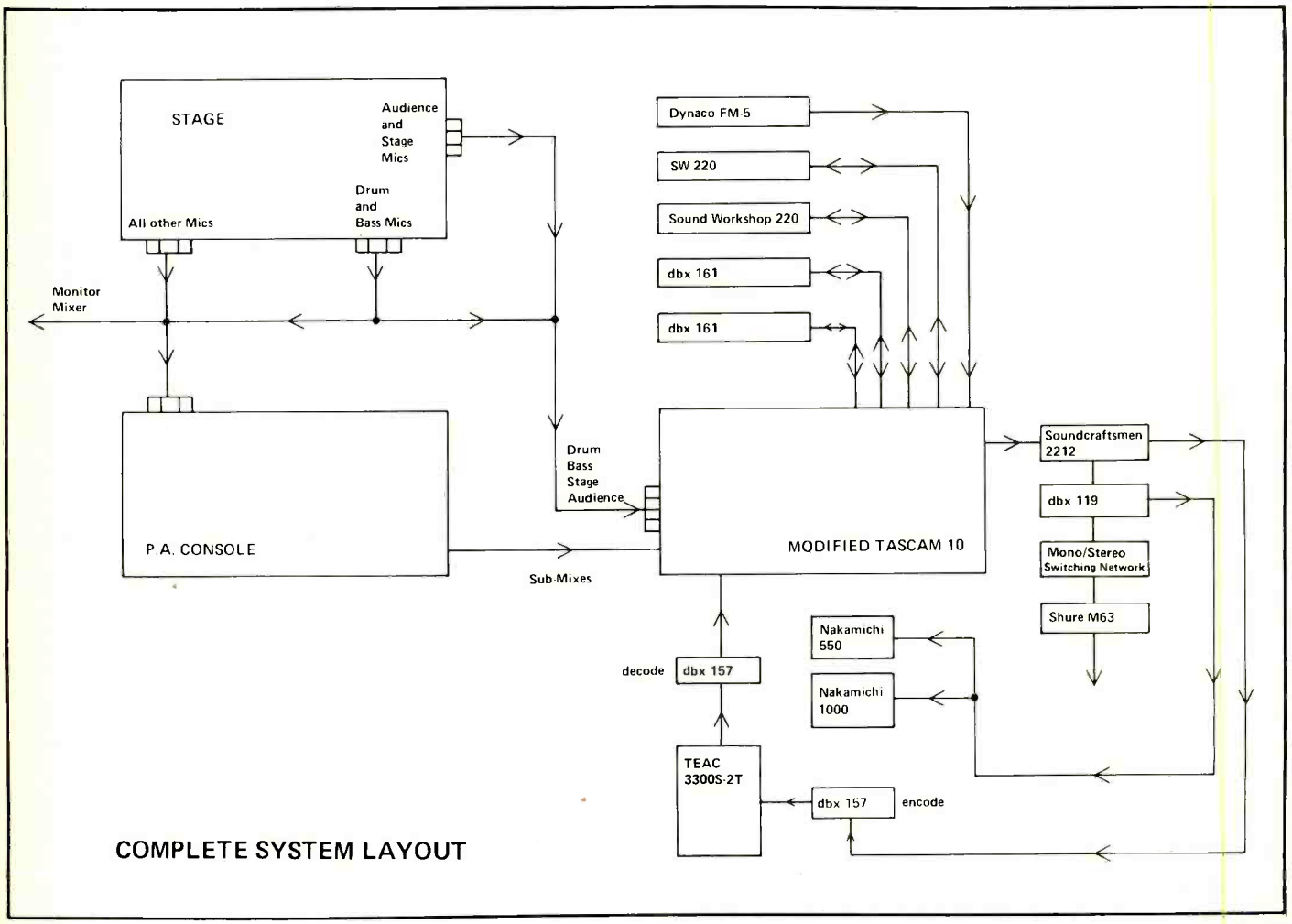
At 7:45 P.M. the phone lines have been given their final check, a tentative drum mix has been established. All sub-mix feeds from the P.A. board have been checked (only that they work has been verified, levels and "the mix" will be set after they're "on the air").

The cue is taken directly from the tuner in the van and we're "live" on the air with interviews explaining the One-to-One Foundation and Chicago's involvement. The interviews conclude and the station returns to regular programming before the concert begins. The interview segment provided an "on the air" check for the console and phone lines. So far so good, but four Electro-Voice RE-15s placed around a conference table is a far cry from thirty-five mics and Chicago "live" on stage.

At 8:20 P.M. the cue is taken from the stage, transmitted to the van via a headset communication system, and then on to the station DJ through the extra phone hook-up to interrupt the song on the air " . . . and go to the Nassau Coliseum, for Chicago, "live" on WLIR-FM." The on-stage introductions were completed and the group opened strong; the tentative drum mix was right-on, and the instrument sub-mix was quickly brought in. The first vocal entrance came in smoothly, and soon the horns and percussion followed. Not bad for flying blind. By the end of the first song all was in order, and it was time to check the EQ and compression on the phone-line feed, based on A/B comparisons between the monitor bus and the "air" (from a Dynaco FM-5 tuner). Compression was fine, but it was necessary to trim the low end "just a touch" on the Shure M63s. Mono compatibility was checked since many listeners would not be listening in stereo.

The first set was over at 9:10 P.M. and the scene switched to the interview set (beside the van) for comments from Geraldo Rivera and Chicago group members. Twenty-five minutes later we're back on the air with Chicago until the final encore is finished at 10:55.

Chicago and all involved had gotten through the "live" radio broadcast by giving a great performance.



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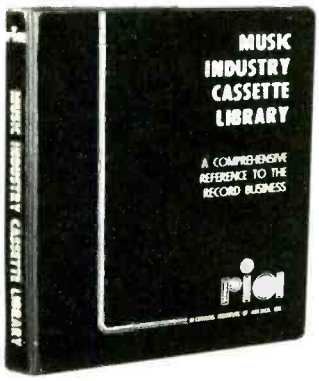

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...and

Sound Re

It started like a good news/bad news joke.

GOOD NEWS: Provide a high output sound system for Woody Herman, Herbie Mann, Earl Klugh, Sarah Vaughn, the George Shearing Trio, the Thad Jones/Mel Lewis Band, the Count Basie orchestra, Ella Fitzgerald and Dizzy Gillespie. Three performers a day for three days with each concert starting at 5 P.M.

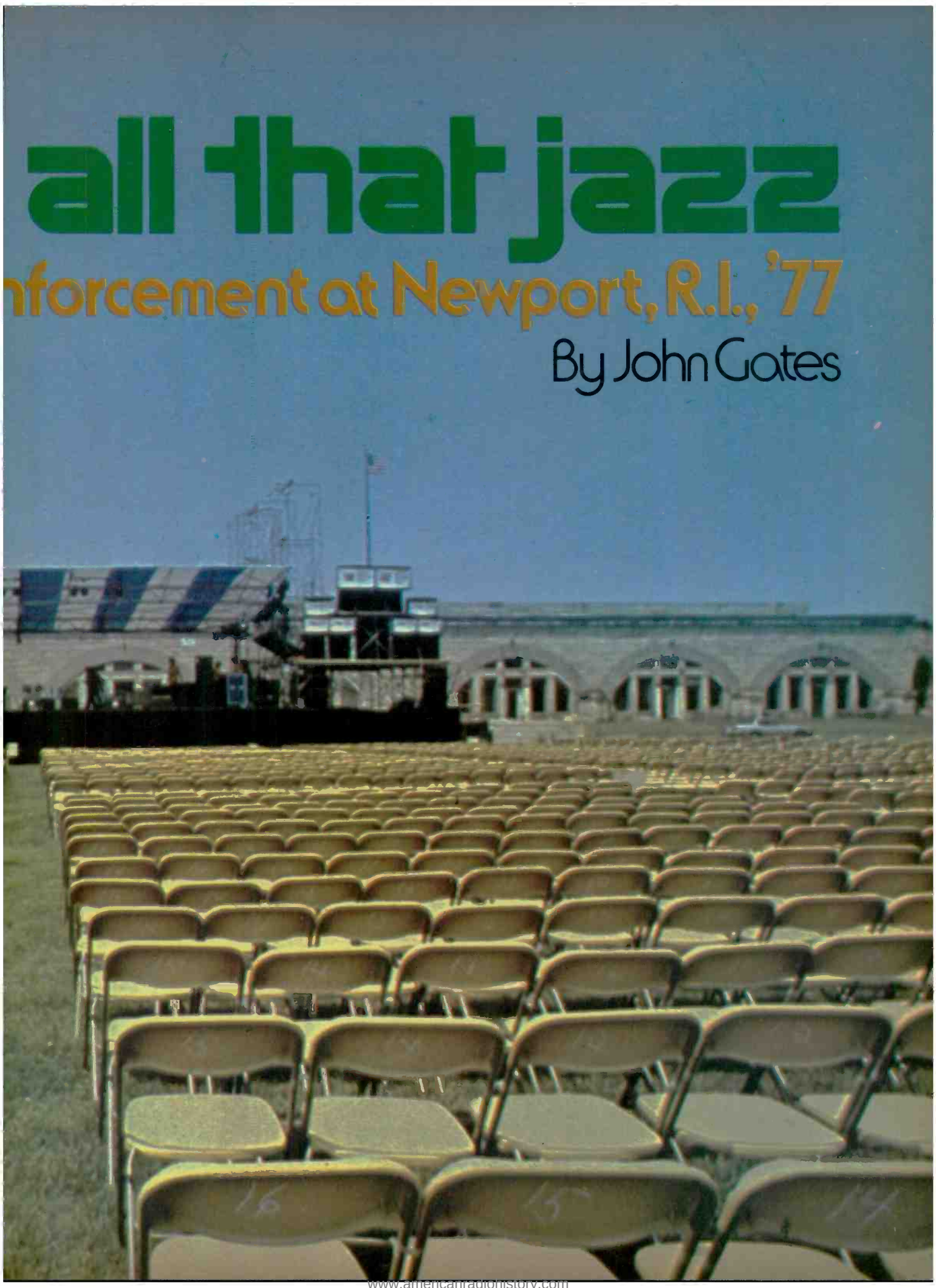
BAD NEWS: The performers would appear during the July 4th weekend (at the height of the tourist season) in the middle of Fort Adams State Park, which overlooks the America Cup Races in the waters of Newport, R.I. The event was "Jazz Returns to Newport, R.I. '77." Here's the kicker: Fort Adams dates back to the American Revolution . . . You guessed it . . . almost no power/electricity/AC/juice. All sound and lighting power would have to be generated. [250-300 amp. 3 phase 4 wire supplied via fully-regulated 150 kVA generator (cummins diesel powered). The generator was located outside the Fort approximately 200 feet from the stage to prevent noise interference. The security lighting power came from the limited AC power available inside the Fort so that the generator condition could not affect security lighting. Total security lighting power—100 amps.]

Capron Lighting & Sound contacted Phylis Adessi, the producer of the concerts, in March. The contracts were approved and signed by early May, so we had plenty of time for planning, equipment preparation and check-out prior to the time of installation at Fort Adams. Because Capron was contracted to provide all the lighting and sound (including the security lighting and paging system), we had a remarkable degree of control over everything affecting our work during the festival.

all that jazz

nforcement at Newport, R.I., '77

By John Gates



Arrival and Set-Up

Jim Warwick, vice-president of Capron, supervised the entire Capron involvement with this festival. He selected sound system components for reliability and performance with an eye to portability and ease of set-up. Like most portable sound systems, Jim designed this system to plug together fast.

Everything in the system is packaged to travel well and withstand various weather conditions—an important consideration when doing outdoor work. The power amplifiers and long-throw horns are housed in custom-built wood road cases. The power amplifier racks are castered; the horn and bass cabinets have to stack, so they are not castered. Custom Anvil road cases are used for all mixers, microphones, microphone cables, microphone snakes and stands for maximum protection against mechanical shock, dust, weather, etc. The mixers operate in their cases with the service areas exposed for fast and easy access. Where feasible, cables on cable reels are built into their road cases.

Jim arrived four days before the festival was to open to personally make final preparations for the sound and lighting installations and to supervise the scaffolding and staging set-up by Ocean State Rentals.

Two days before opening, the rest of the crew arrived. House mixer—Larry "Links" Faherty; monitor mixer—Al Kennedy; light board operator—Paul Iversen; and swing man—Paul "Buster" Horton. They arrived in convoy formation with:

(a) A 22-foot truck with power-lift gate packed with the sound system.

(b) A 14-foot truck with power-lift gate packed with a Hammond B3 organ, and miscellaneous sound and lighting equipment.

(c) A 14-foot parcel van with the stage and security lighting systems.

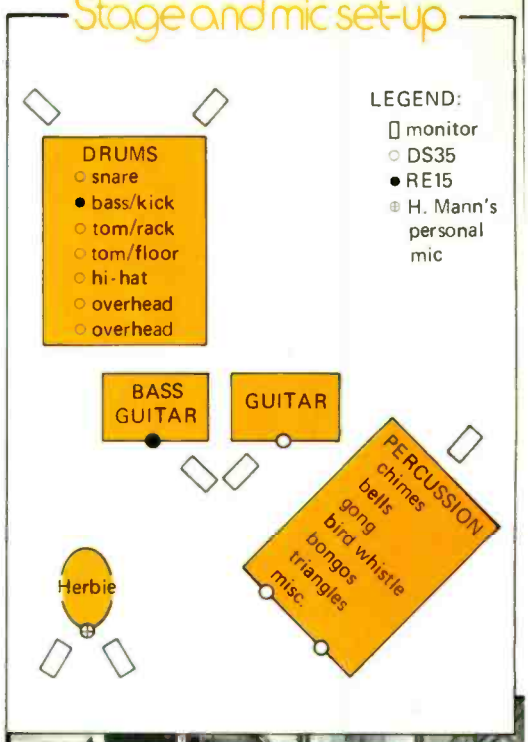
(d) A station wagon with crew gear in tow.

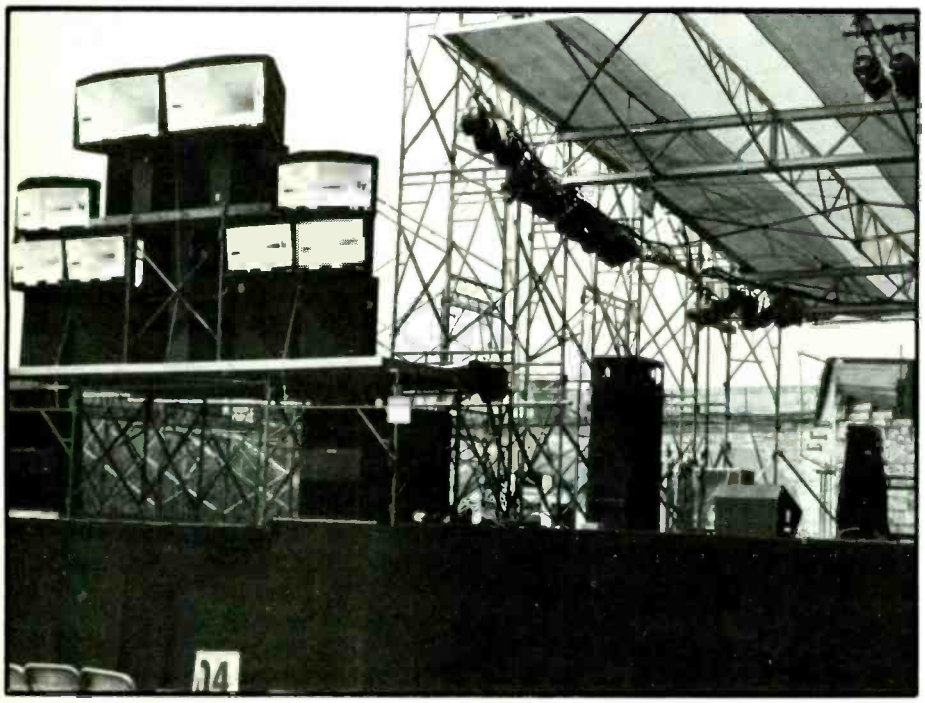
(e) A 20-foot mobile home.

The mobile home was our on-site command center with our only phone

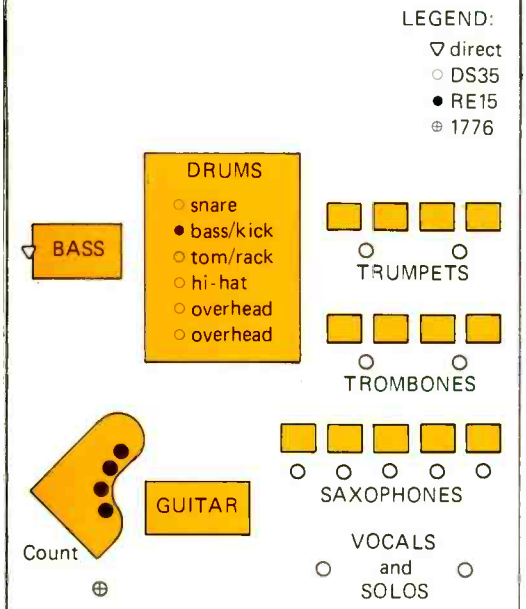
Sound System Equipment List

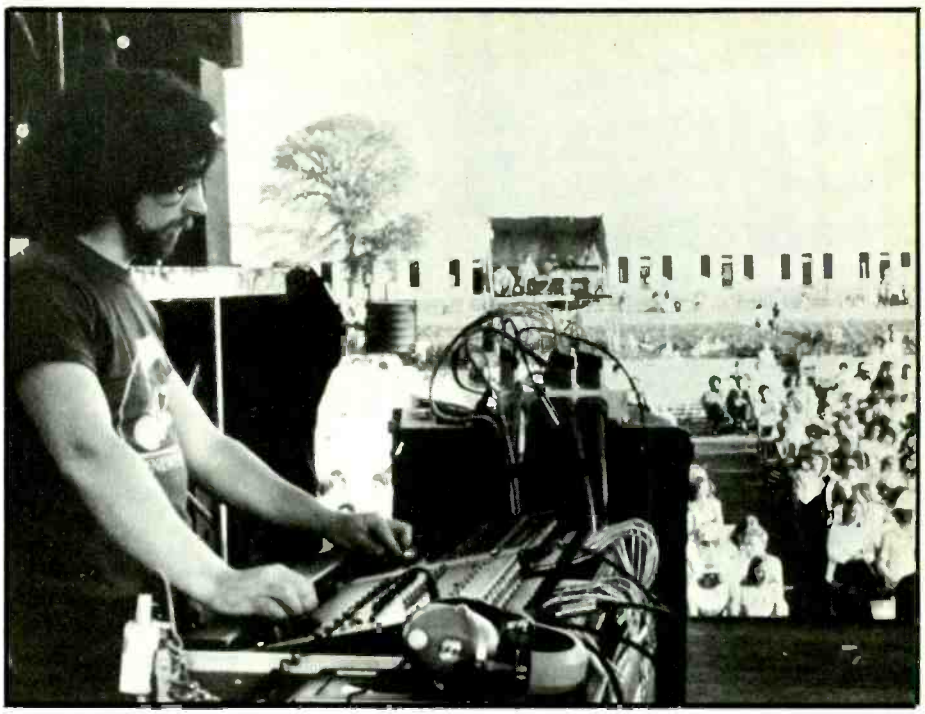
1 Uni-Sync Trouper IV (house mixer)
28 channel with 4 submasters—1 house send, 2 monitor sends, 1 echo send & phantom power supply. Trouper IV equipped with Uni-Sync 241 dual active EQ QLM-1 four-way line amp and limiters and MA-4B LED VU Meters/Line amp package (for sub-masters)
1 Uni-Sync Trouper I (monitor mixer)
18 channel with 1 house send, 1 monitor send and 1 echo send
3 White 4001 1/3 octave active equalizers
1 White 140 real-time spectrum analyzer
1 White 150 real-time spectrum analyzer hand held
1 White 151 pink noise generator hand-held
11 BGW Systems 750A power amplifiers
3 BGW Systems 250A power amplifiers
2 BGW Systems 500D power amplifiers
1 BGW Systems 100 power amplifiers
4 Electro-Voice HR9040 long-throw horns
8 Electro-Voice HR6040 medium-throw horns
8 Electro-Voice HR6040 wide-throw horns
8 Electro-Voice TL806Q Bass Cabinets
8 Electro-Voice TL5050 Bass Cabinets
6 Electro-Voice Sentry IV B full-range speaker systems
2 Electro-Voice Eliminator 2A
8 Capron floor monitors
(With Electro-Voice 12L speakers— cross over at 800Hz)
Electro-Voice 1823M—high-frequency driver
Electro-Voice 8HD—high-frequency horn
1 Captron 31 pair snake





Count Basie
Stage and mic set-up





Photos courtesy of Capron Sound

to the outside world. If you've ever worked on-location, you know how important a phone in a dry place can be. At night, the mobile home doubled as crew quarters. Having the crew housed on job site saves precious time when the nearest hotel is many minutes away (or longer during tourist traffic jams).

Approximately 48 hours were allowed for the set-up and check-out of all the sound and lighting systems prior to the sound check by the first day's performers. This amount of time may seem generous, but it is necessary. This was a first time effort by the producer, and outdoor festivals seem to be governed by Murphy's Law.

Because both entrances to the fort were designed for a horse and wagon, not a modern-day vehicle, all trucks were unloaded at the entrance and the contents were shuttled to the stage area via forklift. The house mixing system was located approximately 200 feet in front of stage left on a 12′×12′×10′ high scaffolding platform. Power amplifier racks were located (in their road cases) at the base of the house speaker stacks on scaffolding wings down stage right and down stage left. The monitor mixing system and monitor power amplifier racks (in their road cases) were located down stage left.

After the equipment was placed in its operating position, the generator was turned on and checked for proper voltage and frequency. Then the power at the main (300 amp) disconnect was checked for proper phasing and voltage. The power that was run from the main disconnect to the power amplifiers and mixing boards was checked also. Then the House and Monitor systems components were connected and powered.

Jim ran pink noise thru the House system (via a mic channel on the mixing board) and the system was checked by Links and Al, segment by segment. When it was determined that the House system was working properly, the House bass amplifiers were set at 80% of maximum output. The House horns and power amplifiers were set at pre-established settings which ranged from 50% to 80% of maximum output depending on the type of horn. Using the pink-noise generator again, the House system was EQed for flat with a reading taken from an RE20 microphone and a spectrum analyzer. After the system was EQed, Jim took spot checks of the sound pressure level at each frequency at various locations in the "house" (all over the fort) using the White 150 hand-held spectrum analyzer. (The system was EQed by ear for the performances.) Al then set the microphones on their stands while Links checked all microphone cables and the mic snake with the Capron Mic Line Tester—MLT. The remote terminator on the MLT allowed Links to check for case shorts, phase reversal, straight shorts and opens as indicated on an LED readout with only one end of the cable plugged into the hand-held tester. Finally, all channels of the House mixing board were checked with microphones plugged in. The Monitor system was checked with much the same procedure as the House system. Sound pressure levels were checked on stage and not in the house.

Microphone List

14	E-V 1776 condenser cardioid
18	E-V DS35 dynamic cardioid
8	E-V RE15 dynamic cardioid
4	E-V RE16 dynamic cardioid
4	E-V RE20 dynamic cardioid

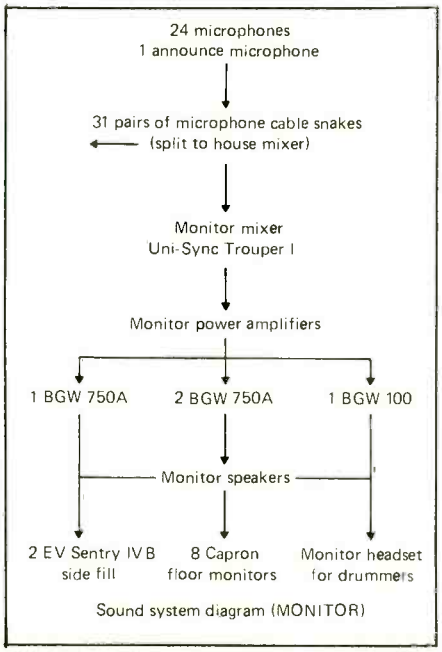
Total: 44 microphones

Operation

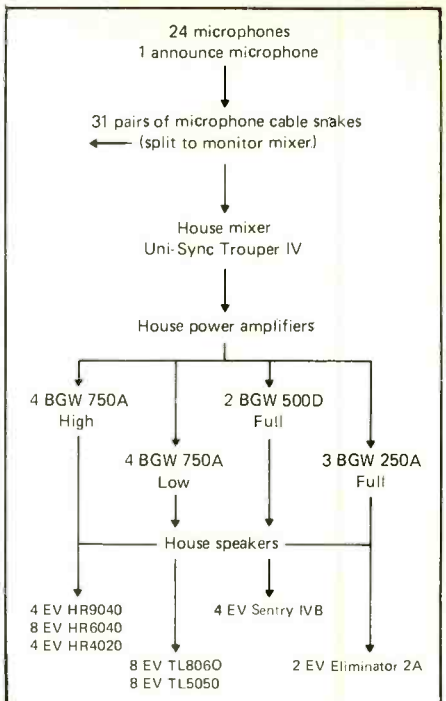
The sound system was ready to go almost 24 hours to the minute before opening. Now, they were ready to deal with the performers, road managers and assorted hangers on. Some performers had sent in microphone lists and set-ups in advance . . . and some performers hadn't.

The performers or the road managers met with Jim and Links to determine the particular requirements for each act. The list of performers indicates a wide range of requirements, music styles and tastes . . . that's what makes it interesting. Unlike rock or pop music groups, none of the performers carried their own sound men, so the House mix was worked out between the road manager and Links. Al worked out the monitor mix with the performers.

Herbie Mann carried a personal microphone to be used for his flute and vocals. During his set, he wanted only the flute in the stage monitors. His set-up diagram shows each musician is miked and each had his own floor monitor. Herbie and his drummer had two monitors each. Contrast the Herbie Mann set-up with the Count Basie set-up. The Count's nine-foot concert grand piano was miked with four RE15 microphones, and the bass (situated) behind the piano was taken direct. The guitar player next to the piano chose not to use any microphone or pickup. Look at the Count's set-up; find any monitors? The Count depends on his ears and eyes to "hear" his orchestra, so no monitors!



Sound system diagram (MONITOR)



Sound system diagram (HOUSE)

It's always interesting to see what seasoned musicians will do in different surroundings with new equipment. During her sound check, Ella Fitzgerald tried the relatively new Electro-Voice 1776 condenser microphone and chose to use it during her set. Both Count Basie and Dizzy Gillespie (on the same bill) took her lead and also used the 1776 for their vocal/solo microphones. During the short, sometimes non-existent sound checks, the solo bus on the Uni-Sync mixers helped Links & Al keep track of everything without potting down all other channels to isolate microphones. The real-time analyzers helped us visually keep track of just what it was we were hearing, quickly and accurately.

The pre-production checkout (before the system left the shop and after it was set-up) was thorough enough to prevent any problems from occuring. It's a very good feeling at the end of three days and over thirteen hours of good music to know that you didn't need any of the backup equipment you brought "just in case." The other half of that feeling is knowing you chose the right equipment in the first place and that you had the right crew with you.

Wouldn't you know, the system had to be struck and packed on July 4th. At least it didn't take as long to pack as it did to set-up. Everything was in the trucks, ready to go home, in just ten hours.

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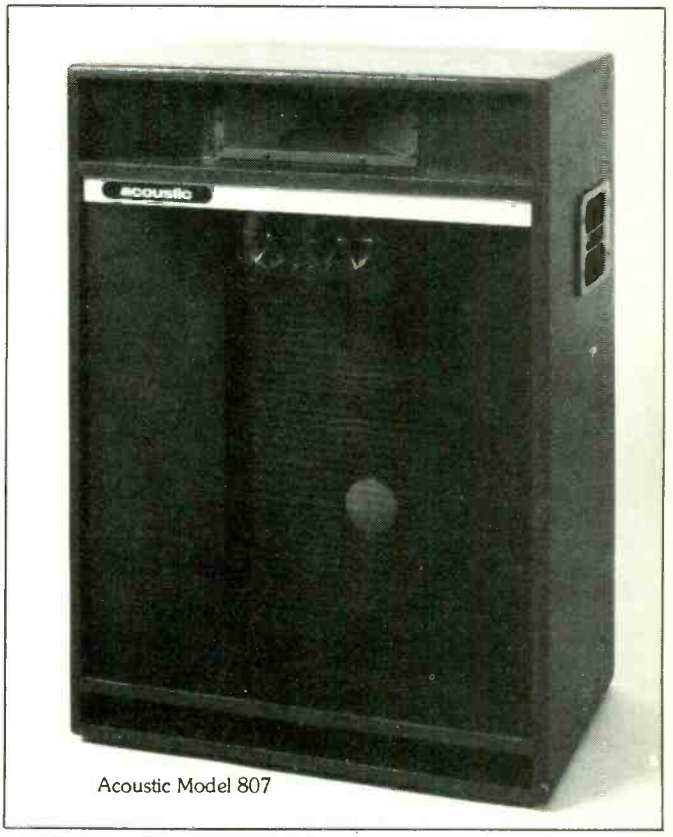

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CIRCLE 82 ON READER SERVICE CARD

Simon & Garfunkel Remembered

an interview with engineer/producer Roy Halee

By Bob Weil

Producer/engineer/record company vice-president Roy Halee has been integrally involved in some of the all-time classic recordings of contemporary music. He got his start engineering Bob Dylan's "Like A Rolling Stone," and moved on to engineer the Lovin' Spoonful and Blood, Sweat & Tears. But he probably is best known for his long association with Simon & Garfunkel—engineering their first few albums and co-producing their Bookends *and* Bridge Over Troubled Water *albums.*

Halee went on to head Columbia Records' San Francisco studios, where he worked with Laura Nyro and other artists before coming to ABC Records in 1975 as vice-president, A&R-Creative. Roy's recent productions have been the critically acclaimed Mark/Almond Band's To The Heart *LP, and a recently released album by a new group, The Big Wha-Koo. The new group is led by David Palmer, lead singer on Steely Dan's* Can't Buy A Thrill *album, and also Carole King's lyricist for the* Wrap Around Joy *album.*

When this interview took place, Halee was just beginning work on an album with a new ABC act—Blood, Sweat & Tears. The album he says is going to surprise critics of the group's recent work.

MR: Roy, how did you get started in recording?

RH: Well, I started in television at CBS and worked my way up into the control/engineering phase of TV before moving to CBS Records as an editing engineer. I spent about a year there editing nothing but classical music prior to moving into the studio end. There I was fortunate enough to start off engineering Dylan's "Like A Rolling Stone," and proceeded from there to do the Lovin' Spoonful, Simon and Garfunkel, Blood, Sweat & Tears . . . all at about the same time . . . so everything fell into place for me, engineer-wise, at that moment.

MR: What machines were you recording on in those days?

RH: Four-track Ampex machines. "Summer In The City," "Daydream," all the early Simon and Garfunkel, and Dylan stuff was four track.

MR: Tell me about the early Dylan sessions. That was a hell of a way to break in as an engineer. Those sessions have an amazing spontaneity, almost like it was a case of "let's just run it through and . . ."

RH: That's just about it. He'd write the song and we'd do it "live." Whenever Dylan was ready we'd just roll the machines because you never knew what was gonna happen. It was almost like doing a remote.

MR: Did you do any overdubbing on those sessions?

RH: None at all, really.

MR: How about the Spoonful? Who was producing when you started?

RH: That was the first thing I did after working with Dylan. Eric Jacobson was the producer.

MR: Was John Sebastian the motivating force?

RH: As I recall, they were *all* contributing heavily and equally, and to me that is the magical thing in the studio — when everybody's into it and it's like a mad house. If someone walked into the studio off the streets, they'd never believe anyone could work like that. But when a group is making great records, there's so much going on that it really is chaos. Jacobson, the producer, was there harnessing this energy . . . trying to keep it somewhat together without holding anything back. Of course, hit records can be made by a formula too, but really creative records are made when everyone's got some input. . . . "How about this?" . . . . "Let's try that" . . . . "I wonder what it would sound like if we" . . . .

The "duh dum *whack*, duh dum *whack*" on "Summer In The City" is just a mic and a sand bag placed in a big garbage can. Crazy, but it worked.

MR: Good things seem to happen when no one's afraid to make a suggestion because they might step on somebody's toes.

RH: Right.

MR: Who was producing Simon and Garfunkel when you first started working with them?

RH: When I first started engineering them, a guy named Tom Wilson was producing the *Wednesday Morning, 3 A.M.* album. He also did the *Sounds of Silence* LP.

MR: As I recall, the version of the "Sounds of Silence" on *Wednesday Morning* didn't have any drums at all while the one on the *Sounds of Silence* album did.

RH: Right. The first version of "Sounds" was actually just a demo, almost an audition; with just Paul and

Artie, and Paul on guitar. We did some overdubs later, when they were out of the country. That version scored and from there on we always used outside musicians. I think Bob Johnson came in to finish up producing the *Sounds of Silence* album and he did *Parsley, Sage, Rosemary and Thyme* and started the *Bookends* album.

MR: So it was during *Bookends* that Roy Halee, engineer became Roy Halee, producer/engineer?

RH: Yes, I finished up *Bookends* and produced *Bridge Over Troubled Water.*

MR: Were you actually producing or co-producing with Paul and Artie?

RH: Definitely co-producing, especially with Paul. Realistically, if you're working with that kind of talent, there's no way it can be anything but a co-production. If the artist hears in his head exactly what he wants then the function of the producer is to get that down on tape, and no matter what the credits say, the artist is co-producing. On the other hand, some "super-stars" really don't have the slightest idea of what they want. Then the producer has to take control in order to do his job—bringing out the best in the artist.

I've never been into the head trip of putting "the Roy Halee signature" on a record. Whatever works best for the record is what's right, whatever's realistic. I really think the "star producer trip" has been blown all out of proportion.

MR: Blown out of proportion by the press? Or by producers who actually get into it?

RH: I think both, man. I think the tyrannical producer is a dying breed. Magical records are a product of mutual respect between artist and producer. I have great respect for Richard Perry—he has the knack, the personality to gain the confidence and get the most out of the artist. And he obviously maintains his objectivity. The same goes for Waronker and Titelman. Peter Asher's another guy who makes marvelously clean records. Bones Howe too; there's an example of what I'm talking about. He's made the transition from making producer's records (the early Association and Fifth Dimension were definitely "Bones Howe records"). Now, the albums he's doing with Tom Waits are fantastic in that they really capture Wait's uniqueness. I think the only great "producer's records" being made today are by the great artist/producers like Paul McCartney and Stevie Wonder. I think Paul's the guy with the most studio chops around. He's not afraid to do whatever's necessary to enhance a record.

MR: Back to Simon and Garfunkel; was Artie doing a lot of the vocal arrangements?

RH: Well, actually, a lot of the arranging was being done by everyone in the studio. Artie, Paul, myself and the musicians.

MR: Which studio musicians were you working with?

RH: Larry Knechtel on keyboards, Joe Osborne on bass, Hal Blaine on drums—that whole crew. When you've got musicians of that caliber you're crazy not to let them work out their own parts. Like Larry has to take credit for the piano arrangement on "Bridge Over Troubled Water." He instinctively knew what to play. On "Keep The Customer Satisfied," that's basically my horn part.

MR: Can you go into the history of *The Boxer*? I've heard it was a long time in the making.

RH: That was my favorite record with them! I found that, as a producer, that was the most challenging and satisfying record I made; there are all kinds of strange things going on. First of all, we started with Paul's great song which truly epitomizes New York City. We cut the original tracks in New York with Fred Carter Jr., who was their guitarist at that point. Fred and Paul had a very magical thing going with their guitars, like they were made for each other.

MR: I've always thought Paul Simon to be a better guitarist than most people realize.

RH: Paul is sensational . . . my favorite acoustic guitarist. I adore the way he plays. He's an even better guitarist than *he* knows. He has a history of listening to everything and everyone, and I think to this day he still takes classical guitar lessons. But, getting back to "The Boxer." It was a gas. After New York we went to Nashville . . . did some overdubbing . . . back to New York for more overdubbing . . . brought in an arranger to write string parts . . . ran out of tracks . . . recorded the strings and some Fred Carter dobro licks on "wild monos" to be sunk in later.

MR: Hold on. "Wild monos?"

RH: You roll a mono, record a part, and worry about putting it in later when you mix.

MR: How difficult was it to sync the parts since no two machines run at exactly the same speed?

RH: Very hard, but not as hard as dealing with the fact that we were working with an 8-track at the time, so on that song we obviously ran out of tracks early along. When I came up with the idea of those voices at the end of the song, and Artie had the brainstorm to record them in a church at Columbia University, I was left with the problem of devising a way to use two 8-tracks in sync—which, at that time, I thought was quite a technical achievement. I was able to do 16-track with two 8-track machines, by using one machine's drive system for the other machine and making sync marks on the tape. We needed at least five or six tracks to put all the voices on, so I made a guide track of the original 8-track to take into this church, did the voices that way, then re-mixed it at the studio synching the two machines together. Not too easy at the time! Which is one of the reasons I think it turned out to be such a truly incredible record.

MR: How about the "bull-whip" percussive sound you got?

RH: That was Hal Blaine's snare; I think we put him in an elevator shaft to get an explosive drum sound . . . sped the tape up a little bit . . . slowed it down . . . heavily limited, heavily compressed—just kept trying things until we got the sound we wanted.

Photo by Bobby Colomby

MR: It must be interesting working with Blood, Sweat and Tears and Bobby Colomby [drummer for Blood, Sweat and Tears] again at ABC after working with them on those well-known albums at Columbia.

RH: The first time I worked with Blood, Sweat and Tears was while engineering their first album with David Clayton-Thomas, the one with "Spinnin' Wheel," "You've Made Me So

Very Happy" and "And When I Die." At that point, Jim Guercio was producing the group. Then I did the next three albums, and Blood, Sweat and Tears went on to work with Paul Buckmaster, Jimmy Ienner and Bob James before coming to ABC.

MR: Tell me a little about your early work with them.

RH: Well, we recorded that first one in a small room at Columbia Studios in New York. At that time, there wasn't a lot of investigation into untested techniques going on, but Blood, Sweat and Tears was very progressive. So, we started playing around with things like leading Katz's guitar through Leslie speakers, recording in phone booths, playing around with EMTs, and I *think* that was the first album with the drums recorded in stereo. I think Bobby will bear me out on that. And even that was such a radical departure from mono drums that we kept them right center and left center.

MR: How about the horns?

RH: Some of them were recorded right there, some were added later.

MR: So you did quite a bit of overdubbing?

RH: We overdubbed some solos and horn parts, but there were no outside players, if that's what you mean. We're striving for the same thing this time. Keeping the sound true to the band—no studio cats. I want to get the input of the band itself.

MR: Well, the new band's a really diverse group of musicians—some have big band backgrounds, some jazz . . . rock . . . r&b . . . the whole musical spectrum.

RH: That's true, and the original Blood, Sweat and Tears were the same way—fine musicians with very different backgrounds. And we're starting to get that same excitement going that I talked about earlier—the chaos with everybody putting their own stamp on the record. If we get that, then we've accomplished something and we'll have another magical record. So far it's going really well. They started off a little tentative, but now they're really playing and saying what they feel. This album's going to have the old Blood, Sweat and Tears "feel," not like their last Columbia album, which sounds like what it is—a bunch of studio cats.

MR: How's the rapport between you and Colomby? Do you have to lay out specific roles?

RH: Definitely not. It's all very spontaneous.

MR: When you're producing do you see yourself as a member of the group?

RH: Absolutely! When you're producing, you have to become a member —a respected member—of the family. If you behave like an outsider looking in, that's what they see you as. A producer should be *the* group member who can also stand back and be objective. But you've got to have the respect and confidence of the musicians.

MR: Aren't you often faced with the choice between going for "perfect sound" vs. grabbing a track that feels right even if its got its flaws?

RH: Sure. The producer's perennial dilemma. Of course the ideal situation is to achieve both, and again, "The Boxer" is the closest I've ever come. Not even knowing what went into it, it just works! You know, you could make a perfect, sterile record, by over-rehearsing, over-recording. Maybe on occasion I've slipped into that perfectionist producer trip, but I'm on guard against it.

MR: How about the problem of musicians losing contact—visual and musical—because of the precautions you have to take against leakage—e.g., baffles, booths, whatever?

RH: First of all, I think leakage can be put to good use. Ask Phil Spector about that. You can hear leakage and say, "Oh my God, I'm in trouble," or, you can say, "How can I use this to my advantage?"

MR: But if you're recording someone who's singing and playing acoustic guitar at the same time, and the vocal's right on but he blows a strum, you could lose a great vocal track.

RH: That's true. So you do a number of takes and splice it up if you have to. You might see fifty splices on one song. Splicing is a viable alternative to punching everything in like some producers do. I do most of my punching on the overdubs after I've spliced up a basic. As a matter of fact, I often recorded Paul's (Simon) vocal and guitar together. The leakage produces a warm interrelationship that you don't get otherwise.

MR: It seems that acoustic guitar and acoustic piano are the most difficult instruments to get a good sound on, but you seemed to get a great sound on Paul's guitar everytime.

RH: Well, with Paul's guitar, I was always experimenting. However, a lot of it is the way Paul plays. If you listen to his recent albums, he always gets a great sound; it's the musician as much as the producer or engineer. Usually I liked using a Neumann U-87 combined with a Neumann KM84. Or if I wanted a very "stringy" sound I'd use a Sony ECM22 with an Electro- Voice RE-20.

MR: How were the mics aimed?

RH: Well, I'd really have to show you, but an example is on "A Song For The Asking." I put a KM84 about two feet back from the sound hole and a pair of U-87s on either side. Sort of like you'd record a classical guitarist to pick up the harmonic structure of the instrument.

MR: And the piano?

RH: Well, I use three mics for the piano, but a lot of people don't use a center mic. But, I like to keep the piano natural in the mix instead of spreading extreme left to extreme right.

MR: So you go for a more realistic panorama of sound—like the musicians might be set up on stage—rather than forgetting that and going for a total aural effect.

RH: Somewhere between the two really. I go for the total sound, but if there's leakage I'll place the instruments in the stereo where they were in the room. Otherwise you sometimes hear something disturbing without even knowing what it is. So, I wouldn't put a floor tom on the far right when it's also being picked up on the left overhead mic.

MR: And what is your favorite miking technique for drums?

RH: Again, whatever works, but some of my favorite mics are AKG 452s, or 451s on the overalls, a KM84, Shure SM53 or even a 452 on a snare. I like a Sennheiser MD441 for the bass drum. I like KM84s very much for the high-hats because they're so clean. Beyers on toms and Sony C500s on the big floor-toms. Just examples you understand.

MR: Do you like heavy compression on the drums?

RH: No. I don't like to use it. I love the drum sound Dudgeon [producer Gus Dudgeon] gets for Elton John's albums. I think they work incredibly well together on everything for that matter, but sometimes it bothers me that it's always the same. It works and it's great, but I like variety!

MR: That's obvious from the album you did recently with the Big Wha-Koo.

RH: Well, the group is so talented. To me, David Palmer's a star singer and writer. They've got three other good vocalists, plus Danny Douma is another incredible writer. They can do

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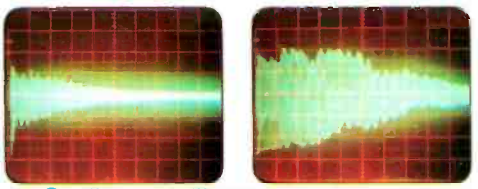

Oscilloscopes illustrate the difference between soft, uncompressed sound (left) vs. loud, compressed sound.

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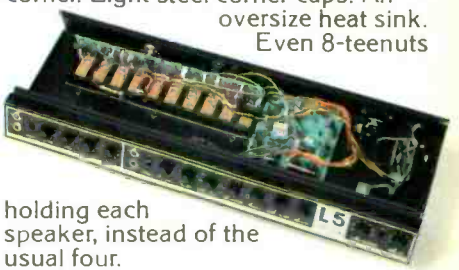

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rock, L.A. rock, ballads, even gospel and reggae—and do it all well. I have really high hopes for those guys.

MR: What mics do you like for vocals?

RH: Again, it depends on a lot of things—the vocalist, the part, the room, the moods, the song . . . a lot of things. So, I might use an E-V RE20 on one kind of voice for an uptempo rock song. Or on a lush ballad with full voice maybe a Neumann U-47. When in doubt, I'll try a good all-purpose mic—U-47, U-87 or U-67.

MR: When you're recording do you go for the "final" sound at the time, or do you record it fairly straight and perfect it in the mix?

RH: I try to get the sound the way I want it while recording. Too many times when you say, "I'll get it in the mix," it doesn't work out, so I try to do my EQ-ing while recording. Within safe boundaries of course.

MR: Do you use a lot of EQ?

RH: You mean am I an EQ freak? If the music calls for it, I guess I'll use whatever I have to.

MR: How do you like to order your sessions?

RH: Always the basic tracks first with the vocalist singing a reference track over the headphones to the musicians. When the basic tracks have been cut and spliced together, a scat vocal goes on the tape. Then sweetening, and at some point along the sweetening process, the final vocal.

MR: Lead vocals before backups?

RH: Vocals are vocals. Usually the lead first, but sometimes all at once.

MR: How about harmonies? Does each vocalist have his own mic? Or do you like the effect of several vocalists around one mic?

RH: I like the blend you get when singers set the balance themselves, singing into the same mic—more power and better blend. Then, as the producer, I can give them some guidance like asking one to step back and another to step up.

MR: How about noise reduction?

RH: Never use it, because I feel it colors the sound. Except occasionally I'll use Dolby's for effect. Like on "Only Living Boy In New York" I put Paul and Artie in an echo chamber with Dolby, and then didn't resolve it, giving it that brittle-brassy sound.

MR: Do you feel you need a lot of engineer's chops in order to be a good producer?

RH: Absolutely not! If you have a good concept of what you want to hear, and you can communicate with a good engineer, you'll come out with a good record. One of the most important things for a professional producer to remember is to let the engineer do his job and to make suggestions without climbing all over the engineer and the board. It's hard for me to relate to producers who do that, because I started as an engineer. If someone's going to get into that, they should engineer their own sessions altogether; which is what I do, because I love it.

MR: Do you run into problems producing and engineering at the same time?

RH: Sure. But it's worth it even though it sometimes gets hectic. Engineering's my right arm, and I particularly love to mix because I feel I can really mix up a storm. Maybe a day will come when I'll want to step back from the board, but right now I wouldn't let anybody mix my records because I think I can mix them better. I remember working in New York as an engineer with Bob Crewe producing. He told me that he usually took over from the engineer for the mix, but he said he didn't feel the need to do that with me, because he felt my mixes were as good as his. That really made me feel good.

MR: OK, tell me about mixing. How much effect do the room and the monitors have?

RH: To me that's always been frustrating man, because I get so locked into a mix in one environment that when I take it somewhere else, it drives me crazy. So I try to find a compromise, but the ideal would be for everybody to hear the record on the speakers I mix them on. And that's not even getting into the loss of transparency that occurs in mastering. The transfer from tape to disc is one of the weakest links in the chain; especially as you get in towards the center of the record and the transients start to go and you lose the top end.

I've actually heard a 12-string guitar disappear from a mix on certain speakers. I guess I do what most producers do. I have several systems at home that are a pretty good cross-section (big McIntoshes, KLHs, JBL 4310s and 10-inch Sonab cubes), so that's what I go by, but I'd still like to mix on each set.

MR: How about mixing a single?

RH: Well, I EQ differently and I mix through the small speakers. If you threw a single mix up on the big monitors, often the voice would be buried.

MR: What goes into being a good engineer?

RH: To me, engineers fall into three categories: Number 1 is the musician-engineer. He takes his cues from a chart and can read music, so he plays the board like a concert pianist; Number 2 is the creative engineer who doesn't have the musical background but is extremely creative, with great instincts for remembering cues, finding the right sound, striving for different sounds and techniques—"Let's try phasing here, feeding that echo through a phaser and delay it, then combine it with a straight feed," always trying something new; Number 3 may be an electronic whiz, but he basically sits there and follows instructions. Of course, some producers prefer that kind of engineer.

MR: And, as an engineer, Roy Halee falls into which category?

RH: I hope I'm a combination of one and two. Creative engineering can snowball into magical records. Example: on "Bridge Over Troubled Water," Blaine was playing "dum-dum, b-dum-dum, dum-dum, b-dum-dum," and I put a combination of a 15″ tape-reverb with a 7″ tape-echo and all of a sudden it was "dum-dum, b-dicka-dah, dum-dum, dum-dum, b-dicka-dah, dum-dum;" and Hal's freaking out because that's what he hears in the headphones, but it's not what he's playing. So, musical ideas can come from innovative engineering. One thing we did for the first time on "Bridge" was cut a super-fast tape-echo on the voices. I'd take a vocal and re-record it going through one machine at normal speed and through the other a little out of sync. So it was like phasing without the oscillation. What's funny is that I heard that same vocal sound on a Beatles' album at about the same time, so we got to the same place at the same time—independently.

MR: When you listen to music for enjoyment, do you find yourself getting analytical? Picking the recording apart and saying, "I would have done this or that differently?"

RH: That's another pitfall I try to avoid, and I really don't let it happen very often anymore.

MR: Where do you see the future of recording going?

RH: There's digital recording and computer mixing, but I'm most intrigued by the video-disc because it opens up staging and whole new areas of production. That's really exciting to me, man. A whole new aspect.

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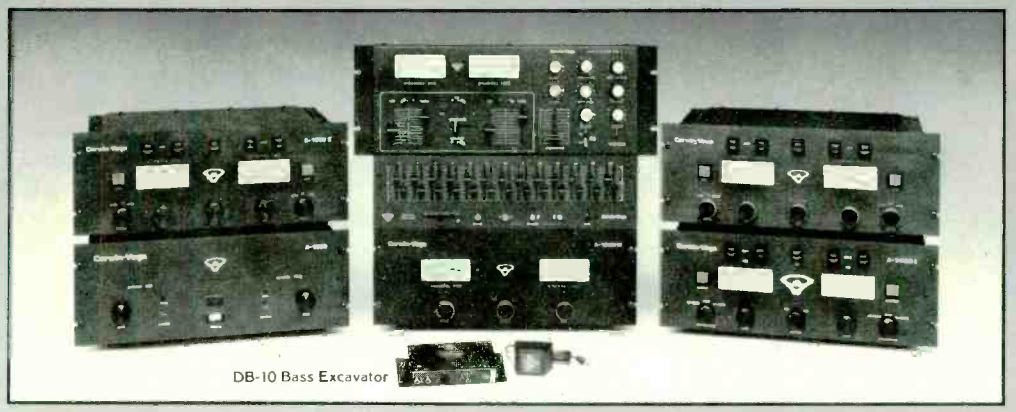

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Ambient Sound

BY LEN FELDMAN

Sub-Woofers or "3-Speaker Stereo"

A couple of months ago, I devoted this column to a discussion of bi-amping and tri-amping. If you recall, bi-amping and tri-amping can eliminate the need for passive crossovers in multi-way loudspeaker systems and are generally thought—despite their higher cost—to provide other sonic advantages.

This month, I'd like to take a look at another trend in speaker system design that seems to be gaining in popularity these days—the use of a sub-woofer (or even a pair of them) to augment the bass response of systems using wide-range speakers which don't quite go "all the way down." Like bi-amping and tri-amping, three-speaker stereo is hardly a new idea. I can remember back a dozen years or more when stereo first came upon the scene in a big way and the listening public's first reaction was: "What, you mean now I've got to have two of those monstrous boxes in my living room?" In those days, the typical popular speaker was a fairly large box (usually floor standing), and the prospect of having to position two of them in a listening room (and having them ideally spaced for good stereo besides) was not greeted with glee even by the most impressed stereophiles.

Two solutions to the problem appeared almost simultaneously. The first of these was the three-speaker approach. Earlier studies had indicated that below a certain frequency, most listeners could not detect direction of sound. Many reasoned, therefore, that one could assemble a satisfactory stereo speaker system by mixing together the bass from the left and right program channels and feeding it to a common low- frequency speaker. Mid- and high-frequencies were kept separated and fed to smaller speaker systems which then had to have only limited frequency response. The "side" speakers, it was reasoned, could then be relatively small units which would not pose as great a problem in positioning as would a pair of large, full-range systems.

In fact, several companies and speaker manufacturers tried just that approach. As most of us now know, the idea did not exactly catch on in those early years, and the "second solution" to the problem fared much better. That second solution, of course, was the development by Edgar Vilchur and others of the so-called acoustic suspension or air suspension speaker design which, though it sacrificed efficiency, did deliver remarkably good bass from a small box.

Why did the first versions of three-speaker stereo fail? For two reasons, I think. First, not enough work or study had been done to determine at what precise frequency we no longer require directional clues. Many people assumed (and some still do) that anything below 200 Hz or so is "non-directional." It is interesting to note that during that same period it was generally believed that high-frequencies (say, above 4,000 Hz or so) were also "non-directional" and that therefore, stereo systems might employ "mixed highs" as well as "mixed lows." At least two proponents of stereo FM broadcast systems which were tested in the late 1950s and early 1960s offered systems in which left and right channel high frequencies were in fact added together to conserve spectrum space.

Extended listening tests conducted during that period quickly revealed that stereo separation was important both at high frequencies and at frequencies well below the 200 Hz point which had previously been hypothesized as being the lower limit of directionality. Again, some of the misconceptions that earlier had been taken as gospel were the result of steady-tone testing. As we have long since learned, testing audio equipment and human hearing characteristics with sine-wave tones bears little relationship to what actually takes place psychoacoustically when our hearing mechanism is subjected to the complex signals which we call music.

Another possible reason for the lack of success of those early three-speaker systems was the failure of their designers to supply a properly designed crossover network to go with them. In some instances, the side speaker systems (those which were fed with actual left- or right-channel information) were allowed to

"roll-off" by themselves at the bass end. While such mechanical roll-off might well provide the required flat response of the total system, with the mid-positioned speaker taking over as the side speakers roll off, power was still being fed from the driving amplifiers to the side speakers, and amplifiers were required to drive rather complex load impedances below system resonance of the side firing speakers. Amplifier power may also have been inadequate for the job and high orders of distortion were produced by the side speakers when they were fed with tones well below their low-end limits for satisfactory sound reproduction.

Coming Full Circle

Well, times have changed, and so has the technology of speaker design. Once more we see a variety of three-speaker stereo system possibilities coming to market. Perhaps the most elegant of these that I saw at the recently concluded Consumer Electronic Show in Chicago this past summer was JBL's newly designed L-212 system. This is an array in which all three speakers of the system are supplied. What's more, the sub-woofer (which has a crossover frequency of only 70 Hz), has its own built-in power amplifier with inputs from left and right channel conventional amplifiers used to drive the "side" speakers as well as the required crossover network.

One of the advantages of the three-speaker system, as pointed out in their excellent demonstration, is that because of the non-directional information produced by the sub-woofer part of the system, that sub-woofer can be placed just about anywhere in the listening room—even well behind the listener. All sound will still seem to be emanating from the plane between the two conventionally placed left and right channel side speakers, which are considerably smaller in size than they would have had to be had they been required to reproduce the full audio range. This is more than a decorating advantage, and should be of particular concern to those of you who face the problem of equipping a control room with monitor speakers or a recording studio area with playback speakers or talk-back speakers.

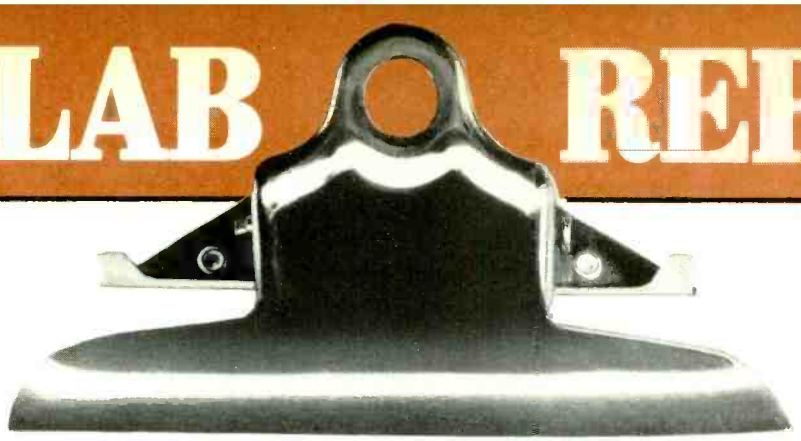
Conventional speaker placement for stereo has always involved two contradictory considerations. In order to achieve a balanced reproduction in the bass range, the positioning of the speaker boxes is limited by the distance between the walls of the room. The final position of any speaker is bound to affect bass response and to create "standing wave" peaks and nulls. This often leads to either overstressed bass or bass that is simply too weak at the listener's position. On the other hand, proper positioning of the speakers in terms of "stereo imaging" and high-frequency dispersion is almost always dictated by the listener's preferred position, so that mid- and high-frequency sounds radiating outward towards the listener are not blocked in any way and so that the listener subtends the proper angle to each of the two side-channel speakers. If one tries to combine these two requirements, we see that on the one hand speaker positioning ought to be variable (for best bass reproduction) while on the other hand it should be fixed to provide a convincing stereo sound pattern.

A properly designed three-speaker stereo system can solve this problem. One can experiment with placement on the sub-woofer until bass reproduction is optimized (without regard to the sub-woofer's position relative to the listener), while the two side-channel units can be positioned for optimum stereo imaging and best dispersion. The sub-woofer, in fact, not only can be placed behind the listener (if that proves to be the optimum point) but can even be hidden behind a piece of furniture or concealed behind a curtain, since such positioning will have no effect whatever on bass energy delivered into the listening room.

Another sub-woofer recently introduced in this country is one made by Visonik, their model SUB-1. Using a 12-inch woofer, this unit crosses over at 160 Hz and is supplied with its own matrix crossover. It is intended for use with that company's D-502 side speakers, updated versions of their David 50 models.

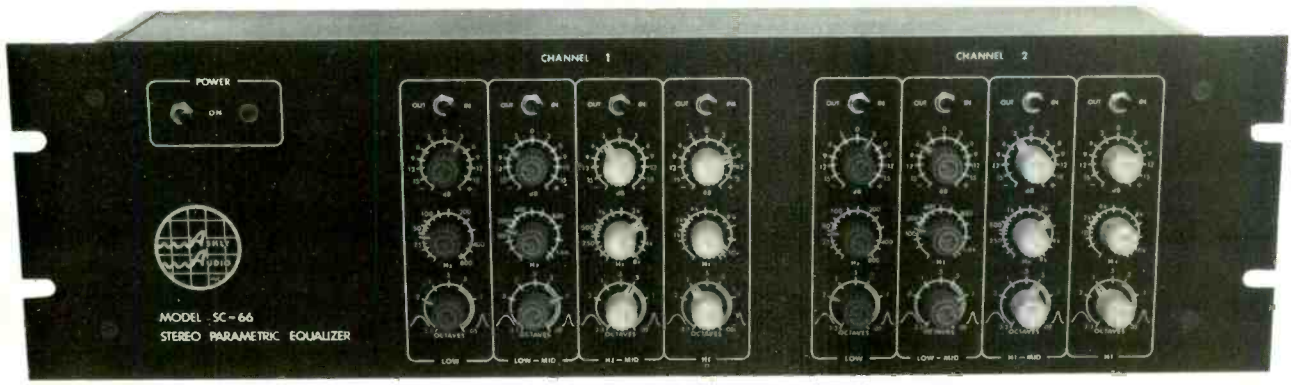
The two companies mentioned are not alone in the re-introduction of the three-speaker stereo concept. For some years now, Janis Audio, located in New York, has been successfully marketing their Model W-1 and W-2 sub-woofer systems which also come with crossover networks and can be adapted for use with just about any high quality side speaker pairs which are then no longer called upon to deliver their very lowest bass.

Readers may also be familiar with the Infinity Servo-Statik 1A speaker system which consists of a sub-woofer cube containing an 18-inch woofer, electronic crossover, feedback sensor and a 150-watt d.c. servo amplifier plus two floor-standing electro-static mid-range modules and tweeter modules with their own built-in power supplies. An all-dynamic three-speaker system has been offered by Phase Linear for some time as well. The bass module of this system, known as the Andromeda II, contains two 12-inch woofers and measures 2′ by 2′ by 2′ while the side speakers contain four 4-inch mid-range drivers and five 1-inch tweeter drivers. No doubt the appearance of these three-speaker systems from such better-known and highly-visible firms as James B. Lansing will prompt other manufacturers to investigate the advantages of three-speaker stereo and to introduce more models in the months ahead. It is interesting to note that while this concept is not new, its present introduction may well be much more successful than were the first attempts to market the three-speaker approach. Perhaps the idea was way ahead of its time back in the 1960s—and it is possible that now, in the 1970s, it is an idea whose time has finally come.



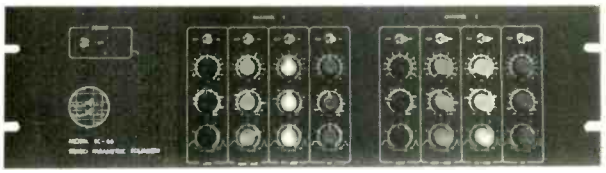
NORMAN EISENBERG AND LEN FELDMAN

Ashly Audio Model SC-66 Stereo Parametric Equalizer



General Description: The Ashly Audio SC-66 is a professional-grade parametric equalizer which may be used as a four-band stereo device, or cascaded to serve as an eight-band monophonic device. Its four frequency bands (per channel) cover "low" (16 Hz to 800 Hz); "low-mid" (48 Hz to 2.4 kHz); "hi-mid" (160 Hz to 8 kHz); and "hi" (480 Hz to 24 kHz). Each band is adjustable in three ways, which is to say in terms of gain or attenuation, in terms of "center frequency," and in terms of the octave-width around the center frequency. In musical terms, these three functions translate to raising or lowering the level of the signal from −15 to +15 dB; moving the center of the group of notes up or down the scale; selecting the number of notes affected from about 40 to a single tone. It is this versatility that is meant by the term "parametric" as contrasted with simpler equalizers which do not permit shifting center frequencies or the width of the band affected.

The front panel controls are logically arranged for the device's operation. Each band has three control knobs. The top knob handles levels and is calibrated from a bit beyond −15 dB through 0 to a bit beyond +15 dB. The middle knob adjusts the center frequency for its particular portion of the total spectrum. The bottom knob adjusts the octave-width from 3.3 octaves to .05 octave, or from a relatively broad peak (or dip) to a narrow spike ("positive" or "negative") depending on how the dB knob has been adjusted. Channel 1 and channel 2 have identical controls and markings. Above each group of three knobs is an out-



Ashly SC-66: Front panel view.

in switch for that particular frequency segment so that equalization can be switched in or bypassed conveniently, as desired. Also on the front panel is the device's power off/on switch and an LED indicator. The panel itself is black; the knobs are colored differently for each frequency band.

The rear contains standard phone-jacks for left and right channel inputs and outputs, and the unit's AC power cord which is fitted with a three-prong (grounding) plug. The SC-66 may be directly rack-mounted.

The non-interacting, but continuous, range of adjustments possible with the SC-66 lend it considerable versatility. Among the suggested uses are control of troublesome acoustic phenomena such as feedback, ring modes, speaker cabinet resonances, equalizing a range of tones or a single musical note. As is true of most equalizers, and especially of parametric designs, the applications are manifold, but the owner's manual gives—as examples—instructions on feedback, correcting for a thin-sounding vocal, compensating for electric bass with poor definition, reducing 60-Hz hum or turntable rumble, correcting for speaker or room resonances and eliminating sibilance.

Test Results: In MR's tests, published specs for the Ashly Audio SC-66 were either confirmed or exceeded except for a minor difference of 2 dBV on hum and noise which could very well be accounted for in terms of differences in test gear. In any event, at −85 dBV and −93 dBV for hum and noise with equalization in and out respectively, the SC-66 certainly is quiet enough for critical applications.

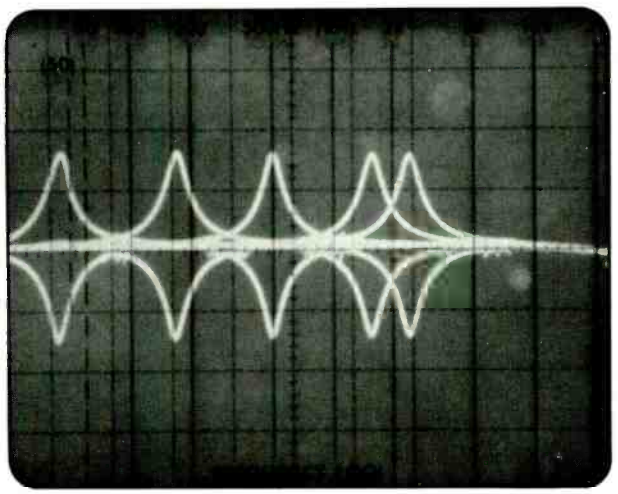


Fig. 1: Ashly Audio SC-66: Low-mid control center frequency can be varied from 48 Hz to 2.4 kHz.

The lab tests confirmed the action and range of the EQ bands and since there is overlap from one band to the next, center frequencies actually can be set anywhere along the audio spectrum so long as four are required (for stereo) or eight for mono. An example of the former is shown in the 'scope photo of Fig. 1. The center-frequency control of the mid-low section was set to its two extremes and to three intermediate positions. Only that single band was used to create each of

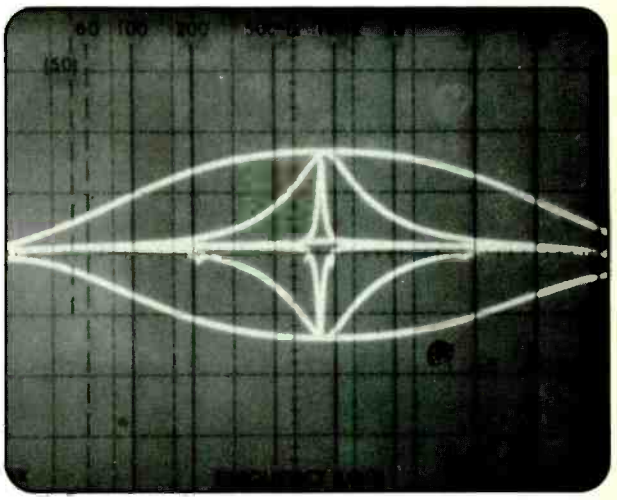


Fig. 2: Ashly Audio SC-66: Bandwidth of each equalizer control can be varied from as wide as 3-1/3 octaves to a narrow 1/20th of an octave.

the five boost/cut response patterns shown, one at a time.

For the test shown in Fig. 2, we set the center frequency at around 800 Hz (using the same low-mid band section of one channel of the SC-66), but this time we varied the bandwidth control from its widest setting (which produces a bandwidth of over 3 octaves), through a medium-width setting, and down to its narrowest setting. In the last setting it is possible to just about attenuate or boost a single note of music without affecting adjacent tones. In sound-reinforcement work this application permits higher overall system gain since specific feedback notes can be sharply attenuated without affecting overall musical response. If the system were used for nothing else, up to four acoustic feedback tones could be suppressed in this manner. And the application of this facility for recording is self-evident.

To further examine the important differences between this type of equalizer and "non-parametric" types, (including those with more bands but with fixed bandwidths and fixed centers), we set up a required response curve that might typically be needed in a large listening room in which a bit of vocalist-presence effect was required, but with attenuation below that frequency because of speaker system distortion limitations, and with removal of a severe standing wave ef-

fect at around 125 Hz (a common problem in many rooms), and finally a bit of treble emphasis to take care of a tweeter that did not quite handle the top treble range above 10 kHz in a linear or "flat" manner. How the SC-66 handled this complex problem is graphically shown in the 'scope photo of Fig. 3. We know of no conventional graphic equalizer that could have produced this overall response curve.

For monophonic applications (probably the most common would be sound reinforcement) the manufacturer states that the two four-band channels of the SC-66 may be cascaded to provide eight bands of EQ. Accordingly, we tested the device in this mode as shown in the 'scope photo of Fig. 4. In addition to suppressing two feedback modes at around 300 Hz and at 2.1 kHz, we still were able to take care of the standing-wave problem, provide the needed bass-boost from 50 Hz to 70 Hz, add the presence-effect for the vocalist at about 1500 Hz and the treble emphasis required by the room. Again, and perhaps even more emphatically, no ordinary graphic equalizer could come close to providing us with this sort of complex response curve.

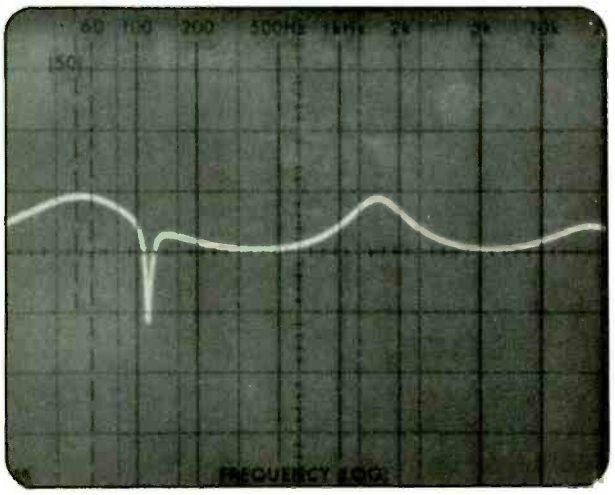


Fig. 3: Ashly Audio SC-66: No conventional graphic equalizer could ever produce this complex response curve.

General Info: Dimensions are 19 inches wide (rack-mountable); 5¼ inches high; 6 inches deep. Weight is 8.5 pounds. Price is $599.

Individual Comment by L.F.: All in all, the Ashly SC-66 seems to be an extremely well-designed unit, precisely calibrated in all its control functions, and capable of extremely wide dynamic range. One feature that might have been useful would be a limit indicator light of some sort for each EQ band, for we did note that when input signal levels are exceeded, distortion rises fairly rapidly. Since, however, the device probably will find its greatest applications in professional sound-reinforcement and studio work, we would presume that adequate metering elsewhere in the system will prevent users from feeding inordinately high signal input levels to the device, thereby avoiding the possibility of overload.

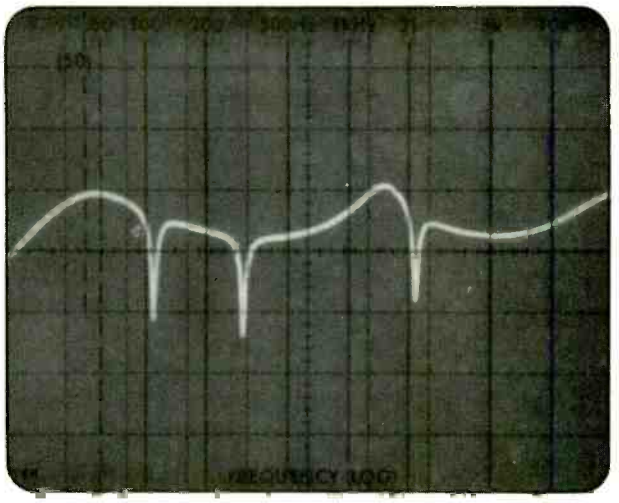


Fig. 4: Ashly Audio SC-66: Used in cascade (monophonically), the SC-66 offers eight sets of parametric controls. It can create even more complex response curves, as shown here.

Individual Comment by N.E.: The SC-66 shapes up as an eminently competent device for professional applications in which it would appear to be outstandingly good for sound-reinforcement work using it as an eight-band mono device, although even for stereo recording work using its four bands per channel, it can do "more" than non-parametric devices with more bands. The avowed intent of the manufacturer in offering this product is to provide a great measure of the versatility of parametric EQ within a format and at a price that could appeal to the budget-minded but professionally-involved sound technician, and it would appear that Ashly Audio has succeeded admirably in doing just that.

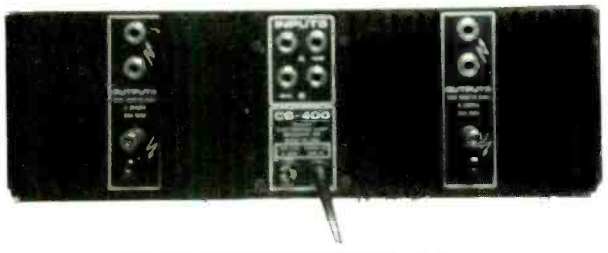
ASHLY AUDIO SC-66 PARAMETRIC EQUALIZER: Vital Statistics

PERFORMANCE CHARACTERISTIC	LAB MEASUREMENT
Amplitude range	±15 dB
Frequency range per control:	
Low	16 Hz to 800 Hz
Low-mid	48 Hz to 2.4 kHz
Mid-high	160 Hz to 8 kHz
High	480 Hz to 24 kHz
	(all end-points within ±5%).
Bandwidth range	3⅓ to 1/20 octave
Maximum input level (flat)	+18 dBm
Maximum input level (at max boost)	+8 dBm
Frequency response (flat settings)	±0.3 dB, 20 Hz to 20 kHz
Distortion at +10 dBm	0.015% at 1 kHz
	0.026% at 20 Hz
	0.040% at 20 kHz
Hum and noise (EQ in)	−85 dBV
Hum and noise (EQ out)	−93 dBV
Gain (controls flat)	Unity ±0.2 dB
Output impedance	50 ohms; terminate in 600 ohms or more

CIRCLE 13 ON READER SERVICE CARD

Peavey CS-400 Stereo Power Amplifier

General Description: The CS-400 from Peavey Electronics is one of a series of power amplifiers designed for commercial/professional use (others in the line include the mono CS-200 and the stereo CS-800). The front panel, of rack-mount dimensions and fitted with handles, contains individual level controls for each channel, marked from 0 to 10. Above each control is an LED overload indicator and below each control is



Peavey CS-400: Rear panel view.

a "line out" phone jack. This jack (on each channel) has a 250-ohm resistor in series with it and so is suitable for driving headphones or to otherwise feed a signal-voltage to some other device. Also on the front panel is the power off/on switch (which glows when activated), and a stylistically-matched high-temperature indicator.

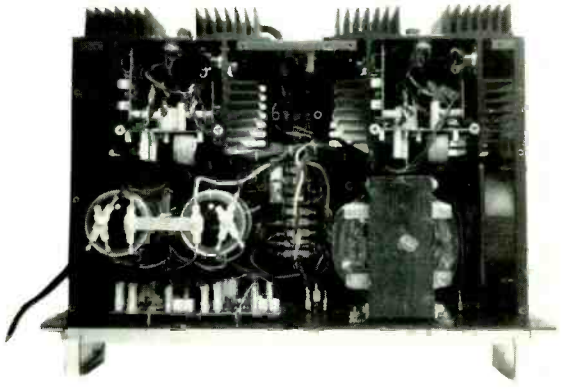
The CS-400 is equipped with a three-speed built-in cooling fan. The control for this feature, a three-position switch, is found on the right-hand side panel.

At the rear are double sets of outputs for each channel, including standard binding posts which also accept banana plugs, and phone jacks. There are double sets of inputs, also phone jacks, so that the amplifier may be fed from two different preamps or control consoles at once.

Rated output is 200 watts per channel for 4-ohm loads (see explanation in "Test Results" section below).

Test Results: Inasmuch as the Peavey CS-400 amplifier is intended for commercial use and is not offered as a consumer-type product, it is exempt from the requirements of the Federal Trade Commission's "power disclosure" rules. Thus, rated power—as Peavey defines it—is given as 200 watts per channel for 4-ohm loads, but the company spells out THD and IM for a 100-watt amplifier working into 8-ohm loads.

To get an insight into just what this amplifier will do, MR tested a few performance areas with both loads, and also observed its behavior at both power levels. The test results indicate that into a 4-ohm load the amplifier will not go into "clipping" until comfortably over 200 watts. Into an 8-ohm load, it clips below 200 watts (at 175 watts). Rated distortion for the 4-ohm load is well below spec for a 100-watt output level. Power bandwidth for a 200-watt level makes it to



Peavey CS-400: Internal view.

14 kHz; for a 100-watt level, to beyond 20 kHz. To those, then, who are mentally "locked into" the prevailing high-fidelity type of spec, the Peavey CS-400 may be thought of as a better than 100-watt per-channel power amp with amazing specs, or a 200-

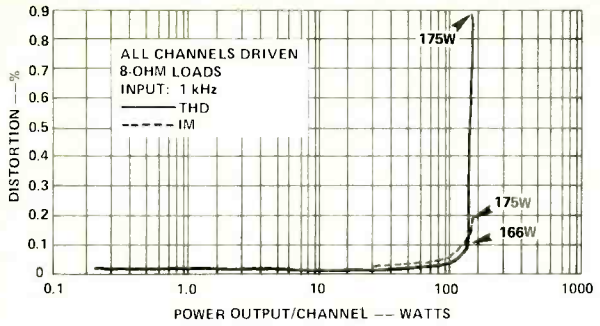


Fig. 1: Peavey CS-400: Harmonic and intermodulation distortion characteristics.

watt per-channel power amp with very good specs but with rising distortion beyond about 15 kHz. Either way, it is primarily a commercial-grade amplifier in terms of its ruggedness of design and construction, and its thermal reliability when subjected to long-term high-power output demands. From all indications, to the best of MR's testing insights, the Peavey CS-400 will fill that bill. Its three-speed built-in cooling fan (which runs silently) is effective: in our lab tests we were aware that we were "punishing" the amp to a far greater degree than would be the case in normal use, and we ran the fan at its highest speed. Even after the

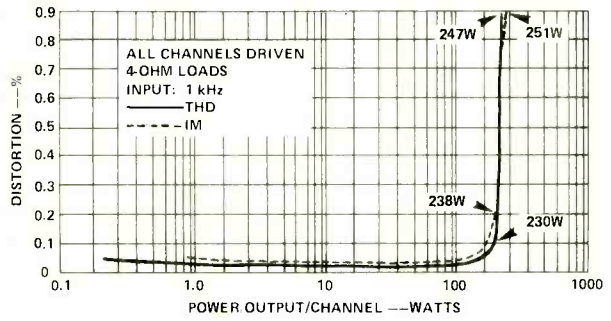


Fig. 2: Peavey CS-400: Harmonic and intermodulation distortion characteristics.

amplifier was called on to deliver near-rated output for the many minutes required to make our measurements, the heat-sink area never really got too hot to touch. And the front-panel high-temp light never had occasion to come on during any of our tests.

General Info: Dimensions are 19 inches wide (rack-mountable); 5¼ inches high; 12⅝ inches deep. Weight is 52 pounds. Price: $424.50.

Joint Comment by L.F. and N.E.: It was interesting to us to note how a commercial amplifier manufacturer lists the performance specs for a product as contrasted to how a high-fidelity amplifier manufacturer must do so in the light of the recently promulgated F.T.C. rules. Were the F.T.C. to have jurisdiction over the "disclosures" for this amplifier they might have a fit. However, they do not, which points up the difference in design philosophy between amplifiers designed for commercial use and those intended for home hi-fi systems. The latter are "spec'd" to appeal to the audiophile, and the FTC's insistence that power-output ratings be accompanied by a statement of frequency extremes over which an amplifier will deliver its rated power at no more than rated harmonic distortion has prompted most hi-fi manufacturers to shoot for a 20-Hz to 20-kHz "power band." Commercial amplifier manufacturers know full well that if THD does tend to rise at the high end (say,

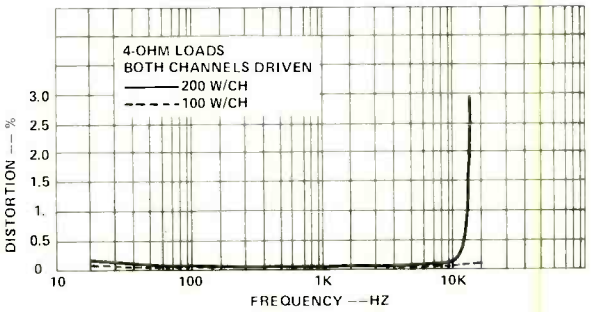


Fig. 3: Peavey CS-400: Distortion vs. frequency.

above 10 kHz), the second harmonic contribution of this distortion will be above 20 kHz, and the more significant third-harmonic THD component will be even further beyond the limits of human hearing. Thus, Peavey seems not at all concerned with the fact that, at 15 kHz, the amplifier can no longer deliver its "rated" 200 watts of power into 4-ohm loads (see Fig. 3)—and, in the case of this amplifier, we're not concerned about that fact either.

What does matter is that the CS-400 appears to be superbly crafted and amply rugged to function in its intended role, primarily as a sound-reinforcement amplifier. However, this is not to say that it could not also serve in home use for stereo systems.

PEAVEY CS-400 AMPLIFIER: Vital Statistics

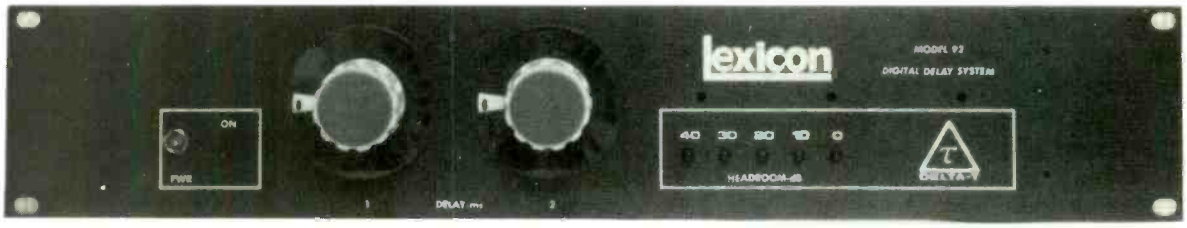
PERFORMANCE CHARACTERISTIC	LAB MEASUREMENT
Continuous power per channel (at 1 kHz)	into 8 ohms, 175 watts before clipping into 4 ohms, 230 watts before clipping
Power bandwidth	into 4 ohms, 200 watts, 20 Hz to 14 kHz into 4 ohms, 100 watts, 20 Hz to beyond 20 kHz
Frequency response	either load, +0, −1 dB, 9 Hz to 30 kHz
Damping factor	into 8 ohms, 120; into 4 ohms, 60.
THD	into 8 ohms at 100 watts, 0.0052% into 4 ohms at 100 watts, 0.0080%
IM distortion	into 8 ohms, 0.015% at 100 watts into 4 ohms, 0.022% at 100 watts
Residual hum and noise	98 dB down at 200 watts output
Input sensitivity	0.83 volt into 4 ohms for 200 watts output
Power consumption	800 watts maximum

CIRCLE 15 ON READER SERVICE CARD

Lexicon Model 92 Digital Delay System

General Description: The Lexicon "Delta-T" model 92 is a digital delay system housed in a low silhouette metal case suitable for rack-mounting. It has one input and two outputs. Time delay for each output is adjustable on separate front-panel controls. These are large knobs that provide stepped adjustments. Each knob is calibrated from 0 to 120 (delay milliseconds) in steps of 7.5 milliseconds. Also on the front panel are the power off/on button and its LED indicator, and an LED display of headroom (in decibels) with five indicators from 40 to 0 in steps of 10 dB. In addition to the delay-time selectors, there are individual screwdriver adjustments for input sensitivity and output level. These are accessible through small holes on the front panel. Connectors at the rear are Switchcraft XLR (three-pin) types, the input being a female and the two outputs being male. The power cord connector, also at the rear, is a three-prong grounding type and a suitable power cord is supplied with the unit.

Basically, the model 92 is a compact, lower-cost application of larger and costlier Delta-T systems from Lexicon and is intended for use in sound-reinforcement work and in studio"/"live"-recording situations. Supplied with the device are detailed instructions and two applications booklets that delve deeply into the use of time delay for sound-reinforcement and in-studio work. Included in the former category are such topics as distributed loudspeaker systems, mic-to-talker distances, outdoor installations, etc. The latter booklet goes into time delay as applied to room acoustics, reverb, image placement, doubling, ambience enhancement, delayed echo send, flanging, etc. In addition to this generous literature offering, the device also comes with a seven-inch, 33⅓-rpm disc recording that contains musical excerpts demonstrating various studio application effects.

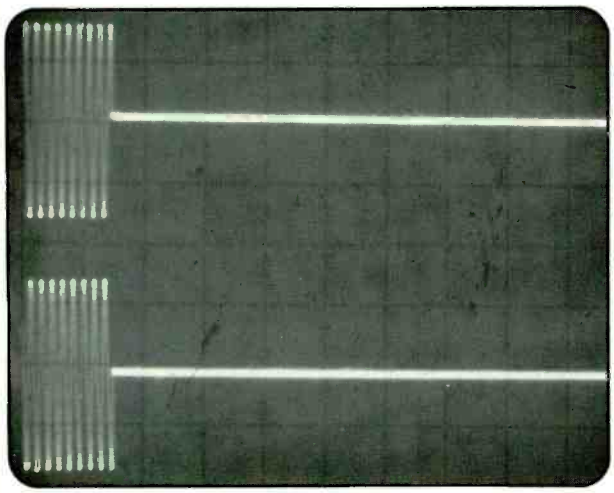


Lexicon 92: Front panel view.

Test Results: In MR's tests, the Lexicon 92 met or exceeded its published specs, and performed very

much "as advertised" which is to say, splendidly. To check out some of the delay parameters in the lab, we introduced a tone-burst signal of suitable amplitude and frequency to the input of the model 92 and then used no delay, then a 15-millisecond delay, and then a 45-millisecond delay. The results are shown on the three 'scope photos we made at the unit's output. These, and additional, tests confirm the device's action, although if you want to be absolutely precise about delay times, add 0.17 milliseconds to the indicated settings.

To measure harmonic distortion and noise at the output, we used a spectrum analyzer which is a better procedure than using a single-meter analyzer. For this test, a l-kHz signal was applied at a level just below the "limit," (as indicated by the O-dB light on the front panel being extinguished), and swept the filters of our analyzer from 20 Hz to 20 kHz. The results are shown on the fourth 'scope photo, with the desired l-kHz output signal represented by the spike at center-screen. Each vertical division here is an amplitude of 10 dB. Thus, 2nd harmonic contribution is down about

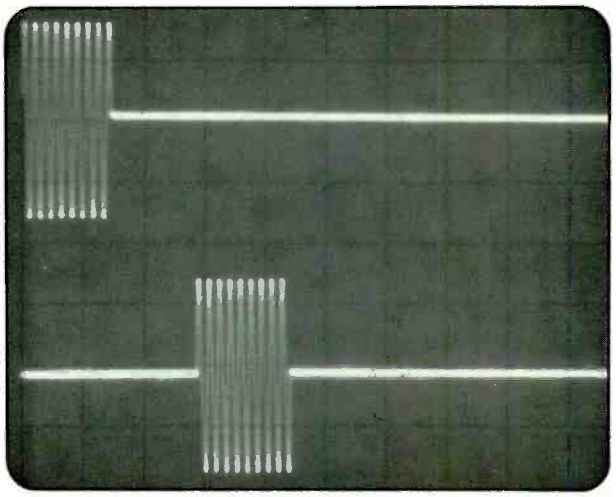


Lexicon 92: Upper trace shows input tone burst; lower trace is output delay of 0 milliseconds.

72 dB (with respect to the fundamental), while the 3rd harmonic content is down some 64 dB (equivalent to 0.063% distortion). In MR's view that is near-amazing performance considering that the signal had been first pre-emphasized, then translated to 12-bit digital code, stored in memory and translated some milliseconds later back to analog form.

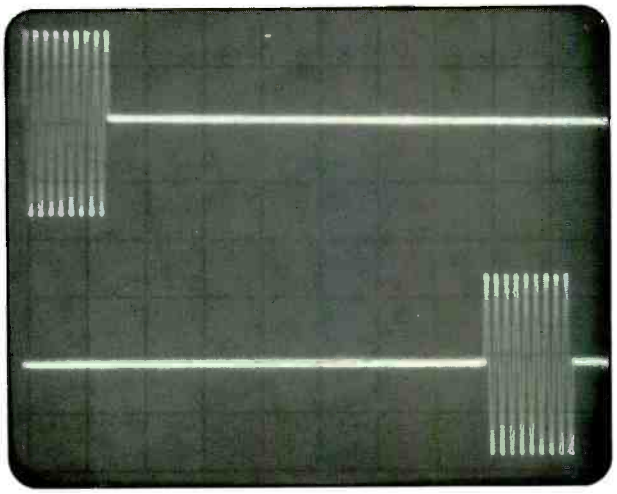
General Info: Dimensions are 19 inches wide (rack-mountable); 3½ inches high; 11½ inches deep. Weight is 8½ pounds. Price: $1560.

Individual Comment by L.F.: As I see it, Lexicon's model 92 may be regarded as their model 102's little brother. It offers only time delay and none of the special effects (resonance, flanging, etc.). Though Lexicon mentions its possible use for echo send, ambience enhancement, and double-tracking use in studio and "live" performance situations, I suspect that the model 92 will find its greatest application in sound reinforcement where zoning of speakers is becoming



Lexicon 92: Upper trace is input tone burst; lower trace shows output delay of 15 milliseconds.

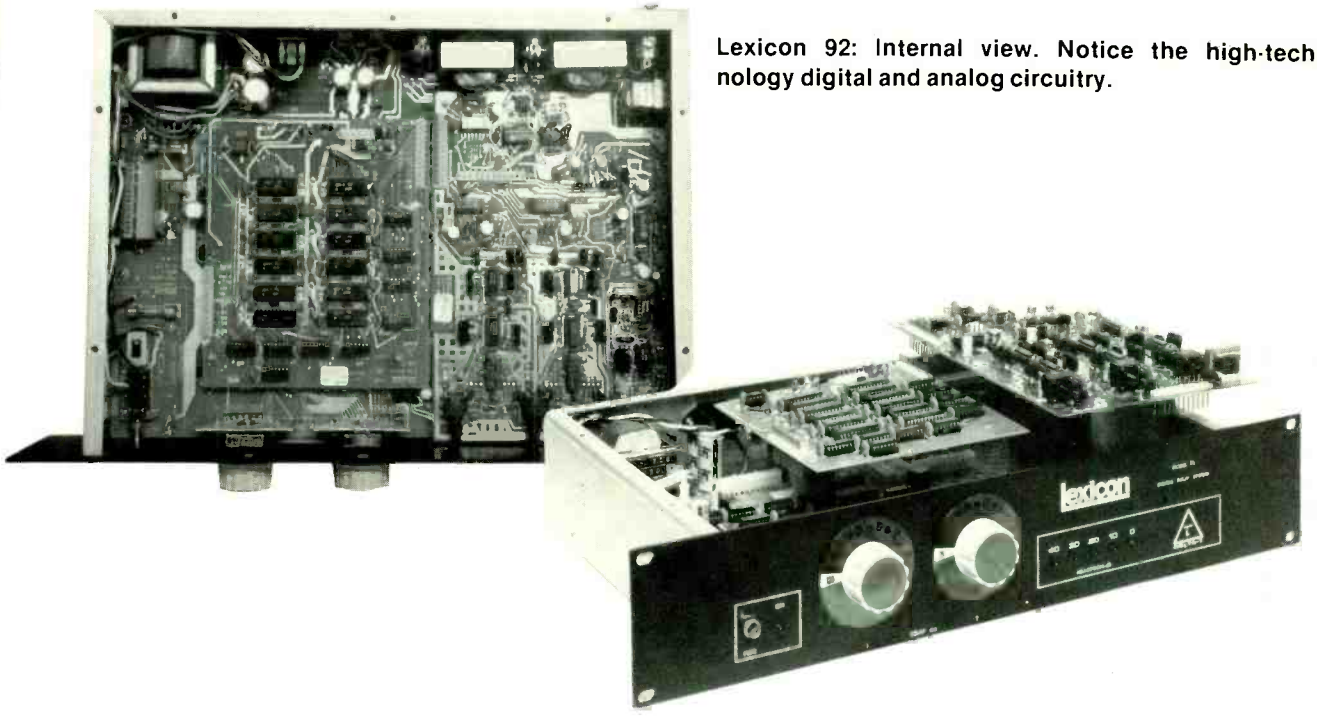
increasingly important for well-designed auditorium and concert-hall sound systems. The model 92 is, after all, a mono unit whose two outputs provide identical signal content, albeit at different adjustable time-delay increments. As such, it makes an ideal delay unit for smaller installations where two levels of delay for speaker zoning are found to be adequate.



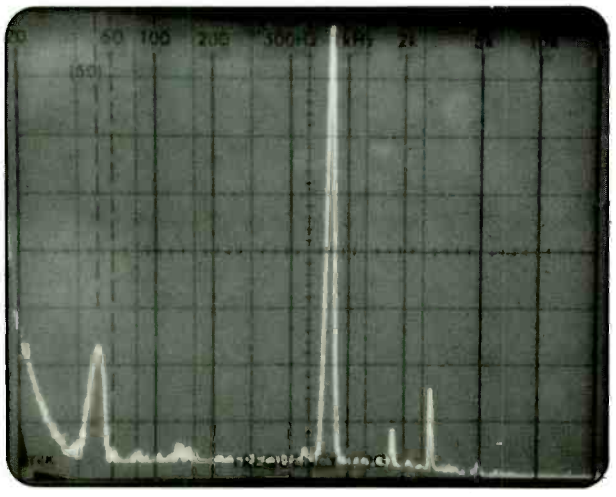
Lexicon 92: Upper trace is input tone burst; lower trace shows output delay of 45 milliseconds.

While my comments are usually confined to a device's performance. I must in this instance make mention of the extremely sophisticated circuitry used by Lexicon in this unit. Input signals are sampled

34,133 times per second, and are converted into 12-bit "floating point words" for each of these samplings. Ten bits are used for the mantissa, and the remaining two for an exponent. Had straight binary encoding been used in the A/D system, 15 bits would be required to encode the same available dynamic range, and that would have necessitated considerably higher cost.



Lexicon 92: Internal view. Notice the high-technology digital and analog circuitry.



Lexicon 92: Third harmonic distortion of "delayed" 1 kHz output signal at just below "limit" was 64 dB below reference output level (0.063%); noise level was considerably lower than that.

Individual Comment by N.E.: The calibrated light indicators are a welcome feature on the front panel. If you have had some experience with digital time-delay units you know that if they are overdriven with too much input signal, things at the delayed output can "fall apart" sonically speaking. With the model 92 there is a full 40 dB of visible indication before that disaster occurs and, since you are able to adjust the "limit input sensitivity" over the range from +8 dBm to +18 dBm, it is possible to take full advantage of the unit's dynamic range capability without constantly worrying about possible overload.

P.S.—If you do nothing else about this unit, try to get hold of the manufacturer's Application Notes AN-2 (for sound reinforcement systems) and AN-103 (for studio applications). They contain a wealth of information about this new technique. And try to listen to that seven-inch disc recording. It's a very good demo of the technique.

LEXICON MODEL 92 DIGITAL DELAY SYSTEM: Vital Statistics

PERFORMANCE CHARACTERISTIC	LAB MEASUREMENT
Dynamic range (ref. limit level)	93 dB, "A" weighted; 87 dB, unweighted
THD (plus noise)	0.065%
Frequency response	+0, −2 dB, 20 Hz to 14 kHz
Delay range	0 to 120 msec, in 7.5 msec steps
Input level at limiting	+8 to +18 dBm, adjustable
Input impedance	10 K ohm min., floating
Output impedance	90 ohms max., transformer coupled
Connectors	XLR-3
Power requirements	100-125/200-250 volts AC, 50/60 Hz; uses 20 watts
Power off/on muting	0.5 sec. on; 50 msec off.

Tangent 1202 Mixer

By Jim Ford and Brian Roth

General Description: One of the latest sound reinforcement mixers available on the market is the model 1202 manufactured by Tangent Musical Engineering. This inexpensive 12 input, stereo output unit provides on each input:

(A) Rotary-type volume control.
(B) A pan pot.
(C) Effects send control with pre/post selection.
(D) Reverb send control with pre/post selection.
(E) Monitor send control with pre/post selection.
(F) Three knob low-, mid-, and high-frequency equalization controls.
(G) Input preamplifier gain control.
(H) Peak overload indicator "LED."

The effects, reverb and monitor send controls have a detent at their mid (12:00) position. Clockwise rotation from the center point causes the send signal to be sent to the appropriate bus *after* the equalization and the volume control ("Post"). Counterclockwise rotation from the detent sends the signal to the appropriate bus

after the equalization but *before* the volume control ("Pre"). No signal is sent to the bus when a send control is in its detented position.

The microphone inputs are balanced, but no transformers are utilized. Instead, a rather novel circuit is substituted for the standard microphone input transformer. Each input also provides an unbalanced input for high impedance microphones or line level sources. Both of these inputs are affected by the input "gain" control.

The equalization circuit is a modified "Baxendall" configuration with shelving action for the high and low frequencies and a broad bell-shaped curve for the mid-range.The equalizer is very similar to that found in top-grade stereo preamplifiers or receivers. The three equalization pots have a center detent at their "flat" position.

The output section contains the usual master volume controls for left and right outputs. Additionally, an auxiliary input control (for tape playback during intermissions, etc.), effects return control and a reverb return control are included for both outputs. The internal spring reverb circuitry features bass and treble equalization controls on the front panel.

The monitor master section provides a master monitor volume control and a five-knob, five-frequency equalizer with graphic-type equalization curves.

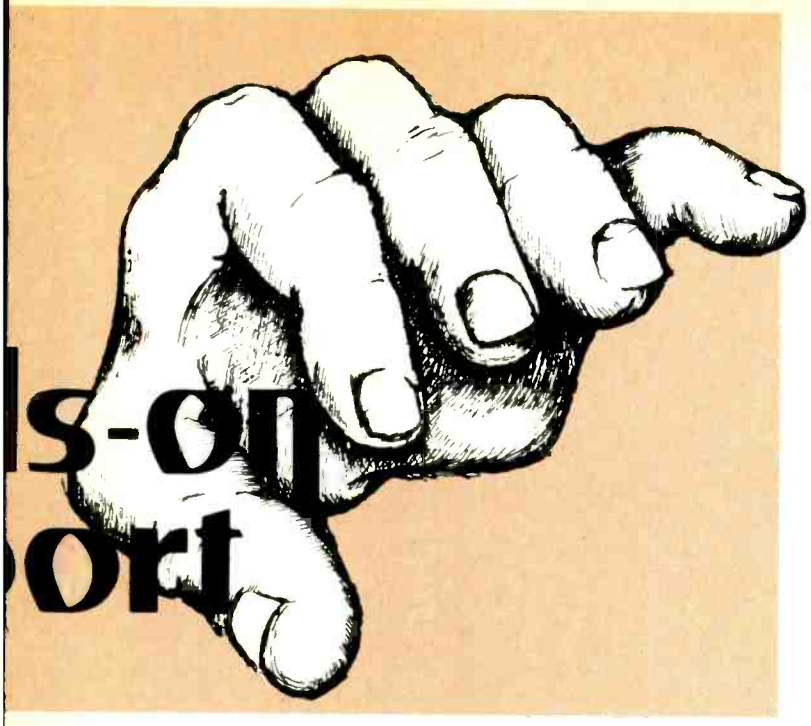
The headphone output phone jack located below the padded "wrist rest" can monitor the main outputs in stereo or the monitor output in mono. A rotary switch for main or monitor source selection and a headphone volume control are located on the front panel.

Metering for the left, right and monitor outputs consists of three rows of LEDs calibrated at −20, −15, −10, −5, −2, 0 and +3 VU. A power indicator is located above the meter LEDs.

The rear panel of the mixer contains all the input and output connections. The main left and right as well as the monitor outputs are available as balanced or unbalanced signals. The reverb send and return jacks will disconnect the internal reverb circuitry if an external reverb unit is patched into the mixer.

All inputs or outputs that are balanced utilize 3-pin "cannon" type connectors. All remaining connectors are quarter-inch phone jacks.

The Tangent mixer is non-modular. Each input or output is wired on a separate printed circuit board that is attached to the front panel by the various "pots" associated with that channel.



Field Test: After donning our orange Tangent T-shirts, we took the mixer to a club that featured jazz artists. The mixer was interfaced with a medium-sized tri-amped sound system.

We all agreed that the unit was very quiet (the ambient noise of the club was much greater than that of the mixer). The bass and treble controls were great; it was possible to make large equalization adjustments without the usual "peaky" sound that the majority of other mixers unfortunately provide. However, we decided that the midrange control affected too much of the upper bass region.

We were disturbed by frequent flashing of the "peak" indicator on the bass drum input, even with the input "gain" control adjusted to its minimum point. We could hear no evidence of clipping, however. Since the jazz band was playing at volume levels much softer than typical rock and roll bands, we questioned whether the mixer would be capable of handling the hot levels found in louder performances. More on this observation later.

The reverb sound was fair; this is typical of most built-in reverb units. Fortunately, Tangent included reverb send and return connectors to allow external reverb devices to be used with the mixer.

The various controls were silky smooth when rotated. Everyone loved their damped "feel." We tended to agree with Tangent's choice of good quality rotary pots rather than the small sliders found on some mixers in the lower-price class. We did miss having slider-type volume controls, but the rotary volume controls created no special operational difficulties.

The headphone output didn't provide very much volume and it seemed prone to distort easily.

The center detented controls for the reverb, monitor and effects sends proved to be rather touchy since only half of the mechanical range of the pot is used. Also, when the controls were used in the "Pre" portion of their range, some confusion was caused since the controls operated backwards from the conventional manner. We would have much preferred separate pre-post switches. We also noted that the "Gain" control was rather sensitive to only slight changes; fortunately, the well damped pots reduced what could have been a real problem.

The LED VU "meters" were interesting to watch in action. In fact, it seemed as if they were a bit hypnotic and distracted the soundman's attention. We also noted that the light-type meters tend to become "washed out" when located in bright sunlight. Maybe we are just old fashioned, but we still prefer seeing a needle swinging across a VU scale, particularly if a peak overload indicator is included.

The unit had a very clean sound overall, and we agreed that the mixer offered an amazingly complete set of features.

Back at the shop, we conducted further investigations of the input overload characteristics. We discovered that the "peak" indicators were somewhat premature when they illuminated. However, we were able to overdrive the inputs with strong signals, even with the "gain" controls at minimum. Using the equalization controls increased the overload problems. We feel that the unit should have included input pads to handle hot signals. Naturally, external in-line pads can be used to eliminate this situation.

We connected the mixer into our listening room setup to conduct further listening tests under a more controlled situation. We concluded that the Tangent was a very respectable sounding unit in terms of low noise and distortion. The performance of the mixer was excellent when its price is considered.

Lab Test: Our first test concerned itself with the input overload characteristics with the equalization controls set at their detented "flat" position; we found that input clipping occurred at +4.5 dBm (1.3 volts RMS). As we noted earlier, this will suffice except for very loud signals (particularly if condensor microphones are utilized). Since the equalizer is located in the circuit after the microphone preamp and before the volume control, any EQ boost will cause a proportional decrease in input signal handling ability. For example, if 10 dB of bass boost is applied the maximum input at low frequencies that the mixer will handle is decreased by 10 dB when compared to the "flat" setting.

The peak indicator would light at about 10 dB before actual microphone preamp/equalizer overload under flat or boost conditions. However, we noted that when the equalizers are set for a high amount of "cut" that the input overload LED wouldn't light under certain conditions, even though the microphone preamp was clipping. Thus, if the bass control was set for full cut and a high-level bass signal was fed into the mixer, the

microphone preamp would overload without any indication from the peak "LED." Although a potential problem, this condition probably wouldn't be encountered very often.

The tables give most of the details of the mixer's performance. We were extremely pleased with the noise measurements. Under all conditions the Tangent was very quiet. As before, we have included a measurement for a "typical mix" situation that is representative of the noise level under actual usage with all 12 inputs activated and a variety of equalization adjustments. This figure is outstanding, indicating that our ears were telling the truth at the gig!

Total Harmonic Distortion vs. Frequency

Frequency	% THD at 0 VU (+dB, 1.25 volts RMS)	% THD at +20 dB output (7.75 volts RMS)
20 Hz	.04%*	.02%*
1 kHz	.008%	.004%
20 kHz	.05%	(see text)

*Mainly hum

Intermodulation Distortion

at 0 VU (1.25 volts) output	.008%
at +20 dB (7.75 volts) output	.004%

Maximum output levels, main outputs

Frequency	High Impedance Load	600-ohm Load
20 Hz	+20 dB (7.75 volts RMS)	+16 dBm (4.9 volts RMS)
1 kHz	+20.5 dB (8.2 volts RMS)	+16 dBm (4.9 volts RMS)
20 kHz	+19.5 dB (7.3 volts RMS)	+15.5 dBm (4.6 volts RMS)

Signal to noise, main outputs, 20 Hz-20 kHz

Unweighted—figures are dB below 0 VU (1.25 volts RMS)

Master volume off	−97 dB
Master volume at normal setting	−80 dB
Typical mix (see text)	−76 dB
Microphone preamp set for 40 dB gain on one input	−79 dB
Microphone preamp set for 60 dB gain on one input	−72 dB
Equivalent input noise	−128 dBm

Distortion measurements were typically masked by the residual noise level. However, once again we were faced with the *slew rate limiting* phenomenon (the inability of the integrated circuit "op-amps" used in the circuitry to reproduce high frequencies at high output levels with low distortion). At 20 kHz, the maximum output level into a high impedance load with .5% THD was 6 volts RMS, or about 2 dB less output than at midrange frequencies. We didn't hear any real evidence of this high-frequency distress during our field test, but, as we've stated before, "faster" op-amps would eliminate any possibility of a problem. The 20 kHz, O VU distortion residual consisted primarily of third harmonic components.

Considering that the O VU calibration point is +4 dB (1.25 volts), we feel that the mixer could stand to have a bit higher maximum output voltage to ensure ample headroom. We noted that a 600-ohm load would cause the output signal level to drop by several dB; this lowered the O VU voltage from +4 dB to +1.5 dBm (.92 volts).

The overall frequency response was quite good—±1 dB from 10 Hz to 65 kHz. We noticed that the equalization controls on several inputs were not quite "flat" in their detented positions; this is probably due to slight inaccuracies in the tracking of the pots.

Since the mixer doesn't use input or output transformers, the square-wave response of the mixer was superlative. No evidence of ringing or instability was observed with the square-wave test. We have seen significantly poorer square-wave respone in mixers costing much more than the Tangent. Additionally, the lack of input transformers reduces low-frequency distortion to a very minimal amount.

Service access to the mixer may be difficult due to the fact that the printed circuit boards that contain the mixer's circuitry are soldered onto the various interconnecting wires. Sockets were not used for the integrated circuit "chips," and it appears that replacing one of them could be a hassle.

The quality of components used in the mixer was adequate. We have had very good experience in the past with the particular brand of pots that Tangent uses on the 1202.

Since the mixer is new, the factory has not completed any type of operating manual; so we cannot comment on its completeness or accuracy. We examined the service information provided by Tangent, and found it to be adequate. However, no printed circuit board parts location diagrams are included (and the parts locations are not labeled on the PC boards). Of course, we are sure that the documentation problems will be solved by Tangent in the near future. It is difficult for the manual writers to keep up with the design and production engineers!

Conclusions: The Tangent model 1202 is one of the best thought out mixers we have seen in its price class. In a number of areas this unit outperforms much more expensive mixers.

Its front-panel layout is easy to deal with, and most all the controls functioned in a civilized manner.

The outstanding aspect of the unit is its sound quality. Distortion was minimal, and the mixer was astonishingly quiet. As we keep saying, the Tangent mixer is capable of higher fidelity than many more expensive devices on the market, and that in itself is an accomplishment.

Considering its overall high performance, it is unfortunate that input pad switches were not included. With this exception, we think that the Tangent 1202 represents a very good value. It includes a multitude of legitimate features and it scores very well in terms of signal fidelity. Put it all together and you have one fine instrument.

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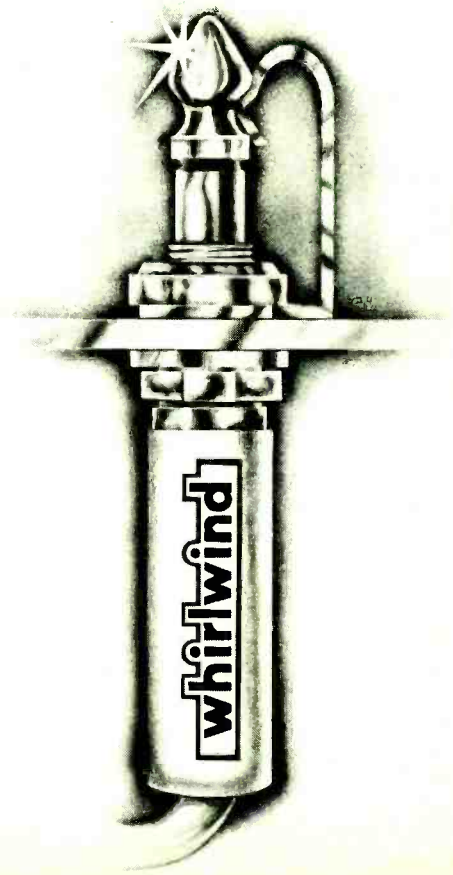


CIRCLE 56 ON READER SERVICE CARD

GROOVE VIEWS

Reviewed by:
SEDGWICK CLARK
CHAS FARRELL-KIMBRELL
NAT HENTOFF
JOE KLEE
GIL PODOLINSKY
HOWARD ROLLER
RUSSELL SHAW

POPULAR

COURTIAL with ERROL KNOWLES: *Don't You Think It's Time.* [Bill Courtial, Steve Whiting, producers; Ken Hopkins, engineer; recorded at Wally Heiders, San Francisco, Ca.] Pipeline Records 2001.

Performance: **Refreshing**
Recording: **Well done**

They're few and far between when you can find a record that features musicians who are capable of living up to their self-labeled musical style(s). Courtial with Errol Knowles can honestly lay claim to the fact that they play a hybrid of jazz, soul, Latin and funk—without turning to disco when in doubt, or more importantly, being steeped heavily in one musical approach and only flirt with the others as would a committed married man. Further, it's so nice again to hear someone sing without sounding like Mick Jagger or Stevie Wonder *and* singing worthwhile lyrics.

Not only is this a debut album for the group but for the label as well, which is cause for real celebration when you realize that there isn't one dog tune on the album. The success here lies totally with the elimination of musical cliches. The Rhodes electric piano is present but not overused, nor is the Elka String Choir (at last, a break from the ARP) merely thrown in to cover a hole. Bill Courtial's guitar is of the more traditional jazz style, which is great in this concept, giving the group sound that change of pace from the routine guitar effects. I especially enjoyed the recording of the congas. They were brought up just right in the mix to add that percussive punch that underlines a feeling and makes the tune work. Everything about the recording was done straight ahead. No gimmickery running away with the production, no recording experimentation done at the group's expense. It is also one of the few group albums where everybody shares center stage both

COURTIAL WITH ERROL KNOWLES. A cause for celebration

equally and often. Finally, this is an album you'll be glad is in your collection to break up the daily heavy-metal assault. G.P.

MINK DEVILLE: *Cabretta.* [Jack Nitzche, producer; Kim King and Ed Rak, engineers; recorded at A&R Studios, New York, New York.] Capitol ST 11631.

Performance: **Average**
Recording: **Reproduced the era**

Mink DeVille first came to the attention of vinyl collectors through a double record set issued late last year by Atlantic Records. It featured the underground punk/revival bands that frequent the CBGB club in New York and was appropriately entitled *Live at CBGB's*. The object of the recording was to see if any one of the groups garnered any favorable public reaction. Apparently, Mink DeVille attracted Capitol's eye and the result of that interest is this album.

To my ear, most of these acts had their roots and inspiration in Bruce Springsteen, for the street-smart lyrics and '50's theme and production are common to all. Those who live in New York or New Jersey probably could profess a chicken-or-the-egg theory, but to the rest of us, Springsteen came first chronologically.

As to the makings of this record, when you have Jack Nitzsche and are going for that late '50's, early '60's sound, how can you miss? The recording utilizes many of the classic approaches and identifiable sounds used to make records during that period—finger snaps, the spotty use of vibes merely to accent, the use of a professional vocal group for back ups (The Immortals), the characteristic reedy sax solo, etc. The material is primarily rhythmic songs with street themes ("Venus of Avenue D," "Cadillac Walk," "Spanish Stroll") and there is much space instrumentally behind the vocals. *Cabretta* really reminds one of all the good things about mono recordings—tightness, simplicity and the excitement/miracle of a good take where everybody played together without mistakes.

The mix is very basic. Drums are primarily snare and high hat with no kick, very infrequent cymbals, and no rolls. The bass is without recognition, heard faintly in the background. No guitar solos to speak of. The use of piano is accidental in effect. In short, there is nothing fancy instrumentally on this record. Vocals are the main intent

MINK DEVILLE: Out of CBGB's and arousing curiosity

here, being clean, centered in the mix without echo or reverb, underlining the lyrical story.

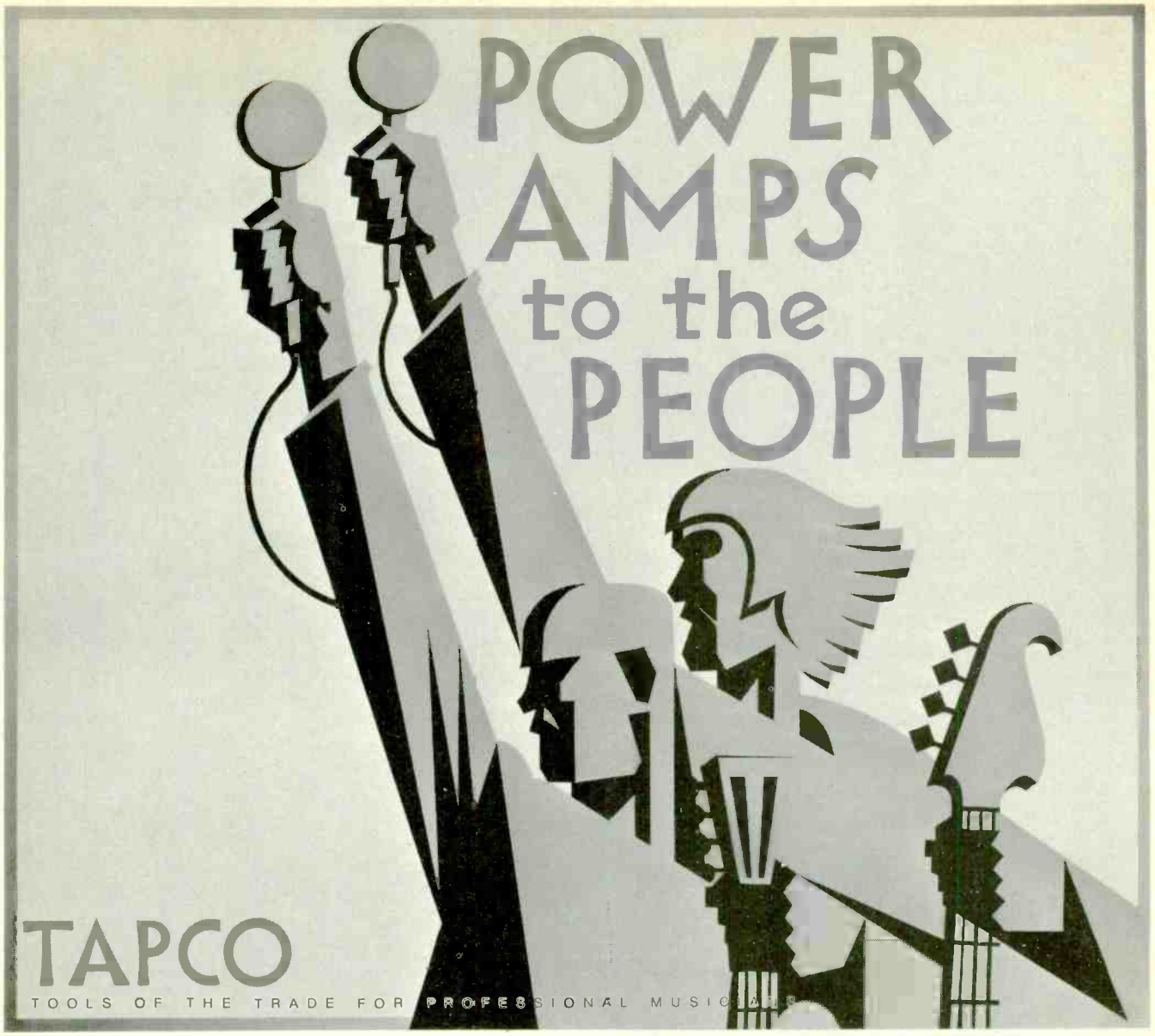
Whether or not this revival sound is to be associated solely with New York or New Jersey, there are a number of groups from that area using this same style. Mink DeVille is not the best nor the worst, merely needing improvement in consistency. For a debut album, it *has* succeeded in arousing my curiosity to see what follows. G.P.

DOCTOR FEELGOOD: *Malpractice.* [Dr. Feelgood and Vic Maile, producers; Doug Bennett, engineer; recorded at Canvey Island, Essex, England.] Columbia PC 34098.

Performance: **Limp as a wet noodle**
Recording: **Not much better**

Musical double standards abound these days. To wit: we're all tired of so-called American boogie bands, repetitive ensembles doing sixth-rate garage imitations of blues standards. This direction is quite unchic.

On the other hand, when you take a bunch of incompetent, out of sync rockers, equip them with British accents, and the right type of hype, they automatically become folk heroes. I can state that back in the southern states where Ah was raised, such bands were the norm. Many aspiring fourteen year old players cut their teeth on stuff like



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CIRCLE 68 ON READER SERVICE CARD

DR. FEELGOOD: A waste of PVC

Muddy Water's "Rolling and Tumbling" (present here). Yet in almost all cases, the groups here in America do it better. That includes ensembles who never made it past legion halls.

Why this sarcastic indictment? Well, to any reasonable, non-affected auditor, Dr. Feelgood has some basic limitations which show next to zero musical ability. The guitar and rhythm tracks are out of time; the lead singer (not identified by name) sounds like a postal worker drunk at a weekend pub, with maybe a slight swagger, but no blues or rock feel. The original compositions are among the most banal, elementary crap to cross these lobes in months. The versions of immortals like the Waters number and the primeval rock stomper "Riot in Cell Block No. 9" convey only a surface bare urgency, not the deep conviction and powerdrive that they were meant for.

Critiquing the production and engineering on this waste of precious PVC is about parallel to issuing qualitative judgement on the performance of the foreman at the sewage treatment plant.

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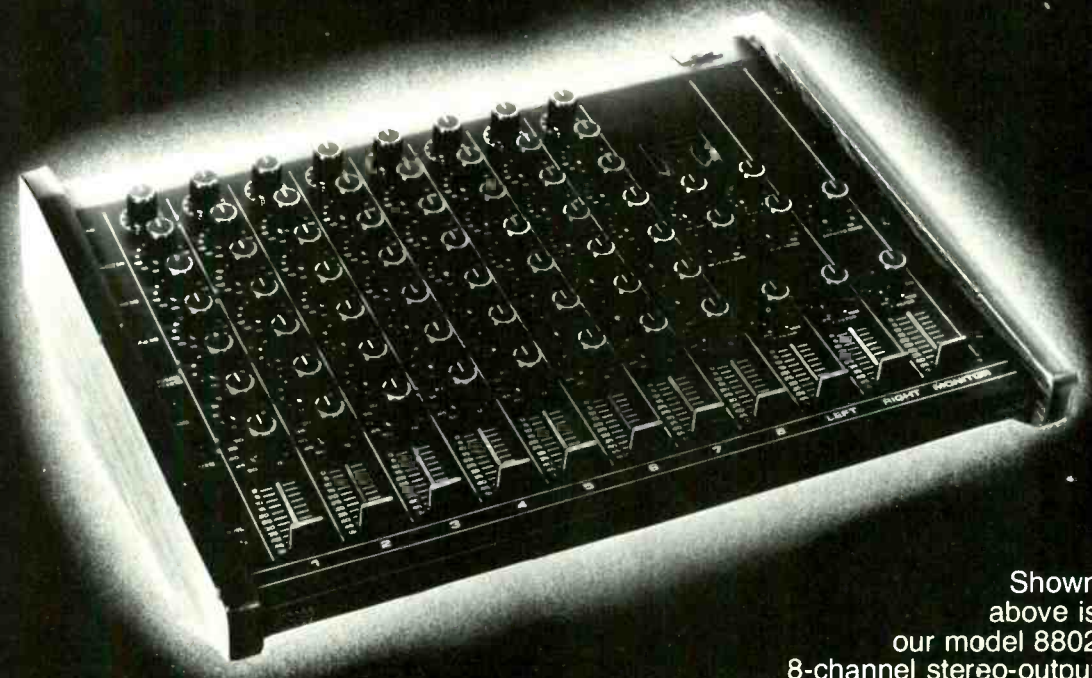

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CIRCLE 96 ON READER SERVICE CARD

Jazz, Small and Large, Brilliantly Distilled

By Nat Hentoff

Quintessence marks Bill Evan's return, if only on records, to a small combo setting—by contrast with the trio dates of recent years. Evans by no means dominates the proceedings, this being an egalitarian collective enterprise as befits the stature of his colleagues: tenor saxophonist Harold Land, guitarist Kenny Burrell, bassist Ray Brown, and drummer Philly Joe Jones. The result is a nearly flawless session of largely gentle, reflective chamber jazz with the musicians sounding as if they had been playing together for many months, so precisely attuned are they to each other's improvisatory sensibilities.

Although everyone involved is quite luminously inventive as they match Evan's acute sensitivity to dynamics, a special tribute is due Philly Joe Jones. A picaresque figure off the stand, Philly Joe, who has slipped in and out of the music scene in recent years, was best known for his steaming, crackling, powerfully witty drumming with Miles Davis in the mid-1950's. But there is also Philly Joe as a marvelously subtle master of graceful, dance-like brush figures —as in much of this set. And softly as he plays, his unerring pulse—the quintessence of jazz—is at the core of the proceedings.

In all, this is the warmest, most thoroughly relaxed (though never lazy) jazz set in months. And the sound of the recording matches the silvery lucidity of the music, with all the parts in delicate but firm balance. Surely much of the non-musician credit for the high standards set by this album is due Helen Keane, the producer who is also Bill Evan's manager. She may be the only jazz manager who herself is so proficient at a control board that she could authoritatively handle the engineering as well as the producing of a date. I've watched her do just that in a mobile unit outside a New York night club.

Someday Helen should have her own label, but that's almost everybody's dream. A jazz afficionado who has actually reached that state of grace is Carl Jefferson, owner of Concord Jazz and producer of many of its albums. The catalogue so far focuses, by and large, on mainstream modern sounds, reflecting the owner's preference for resilient swinging and melodic solos. The sound in the series so far has been invariably clean, brisk and spacious. A case in point is the new *Juggernaut: Frank Capp/Nat Pierce.*

A big band session in the Basie lineage, but with its own distinctly flavorsome character, the arrangements are crisp, with ample space for solo flights by such robust-sounding sidemen as trombonist Buster Cooper and tenor saxophonist Plas Johnson. As a bonus, Ernie Anderson, in some of his most persuasive appearances on record so far, sings and shouts in his uniquely sly, blues-laced way of telling sensual tales. Co-leaders Frankie Capp on drums and Nat Pierce on piano make it all cohere as they joyfully drive this, their very own, big band.

Concord Jazz, Inc. is at P.O. Box 845, Concord, California 94522, and I would suggest you write for a catalogue. Any other small labels that would like to be reviewed in this column can send releases to me at 25 Fifth Ave., N.Y., N.Y. 10003.

BILL EVANS: *Quintessence.* [Helen Keane, producer; Phil Kaffel, engineer.] Fantasy F-9529.

FRANKIE CAPP/NAT PIERCE: *Juggernaut.* [Carl Jefferson, producer; Bill Putnam, engineer.] Concord CJ-40.

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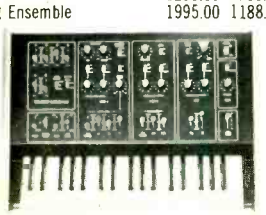

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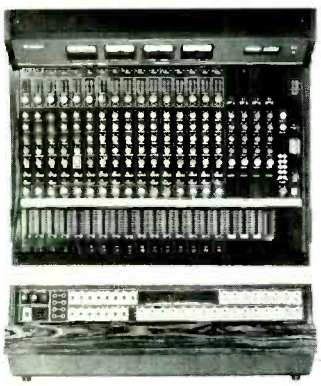

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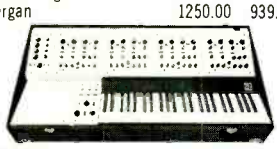

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Both take excrement and try to do their best with it; yet the plant has never been the beneficiary of trendy hype. R.S.

SOUTHSIDE JOHNNY AND THE ASBURY JUKES: *This Time It's For Real.* [Steve Van Zandt, producer; Don Meehan, Ken Robertson, engineers; recorded at Columbia Recording Studios, New York.] Epic PE 34668.

Performance: **Top notch**
Recording: **No complaints**

The key to this recording is its authenticity and sincerity. With the release of the first album one came away with the distinct impression that Southside Johnny was merely riding on the coattails of Bruce Springsteen's success. He was accused of this more so than most because—(1) Springsteen and Southside are good friends; (2) Springsteen contributed material to the first album (as he did here); and (3) Springsteen's guitarist, Miami Steve Van Zandt—who is a long-time friend of Johnny Lyons, having played with him in such infamous Jersey duos as "Southside Johnny and the Kid"—produced the first album as well.

This second album is different from the first, because it succeeds in establishing an identity. I think much of the difference comes from the production concept. I am very impressed that the desire for authenticity in the late '50's, early '60's approach went as far as using such vocal backups from that period as The Coasters, The Five Satins and The Drifters.

The stereo mix is fairly basic. The piano is left, the guitar right, and the vocals, horns, strings, drums and bass center. The exception is "I Ain't Got the Fever No More" where guitar one is left, echoed Rhodes centered with minute fast panning to give a slight definition of movement, and guitar two right. With the exception of the slightly buried guitar solo in "When You Dance" and the sax solo on "Love On the Wrong Side Of Town," this album is void of solos. The four-piece horn sec-

Photo by Ellen M. Gasster

SOUTHSIDE JOHNNY AND THE ASBURY JUKES: Establishing their own identity

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tion of the Asbury Jukes is often countered by the use of a string section. In fact, the presence of the horn section is really what gives the material character and has an effect quite similar to Van Morrison's sound when he was backed by New Caladonia. Overall, this album is very consistent, honest and enjoyable. Music—or recording—needn't be complex to be good. G.P.

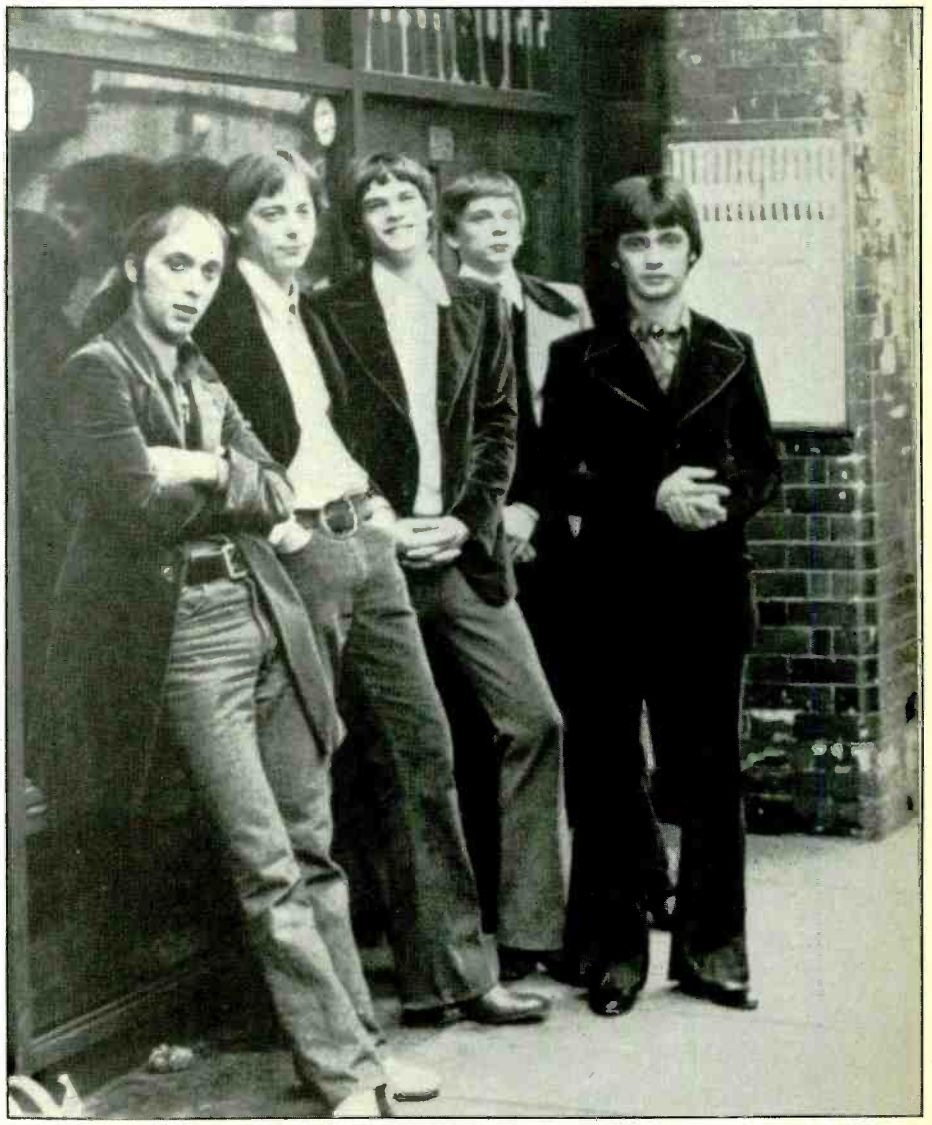
FLAMIN' GROOVIES: *Shake Some Action.* [Greg Shaw, executive producer; Dave Edmunds, producer; engineer not listed; recorded at Rockfield Studios, Monmouth, South Wales.] Sire SASD 7521.

Performance: **Poignantly simple**
Recording: **Pristine clarity**

Oh, for those days of melodic, simple three-minute songs, about non-allegorical teen concerns. Those mid-sixties British invasion pop hits quickly lost their chic, as waves of sociological phenomena made the message muddled and, on occasion, infused pomposity and camp artiste into a previously direct medium.

Call the Flamin' Groovies a bunch of three-chord rockers. Yet, in this instance, the description would be one of both praise and accuracy, for, in truth, this young five-man band from Wales is adept at making the brief, yet crystal-clear, statements that were responsible for the initial British musical waves over here fourteen years ago (has it been that long?).

Produced under the aegis of rocker Dave Edmunds ("I Hear You Knocking"—circa 1973) and famed record collector Greg Shaw, *Shake Some Action* contains fourteen little gems, only one longer than four minutes. Favorites here include the title cut, with an infectious



FLAMIN' GROOVIES: Reminiscent of the mid-sixties British invasion

hook line; "You Tore Me Down," replete with multi-cast Mersey Harmonies; and the acoustically rooted, "I'll Cry Alone."

From a technical standpoint, the production is perfect and flawless. Not that the Groovies project any kind of multi-instrumental virtuosity; three chords seem to be the extent of their active repertoire. Yet within the space of these relatively modest limitations, Shaw and Edmunds have basically followed a no frills, pedestrian approach whereby the guitars are given special emphasis. No great ornate creation, but everything is in place; drums and vocals are given the proper mix; no solos to speak of are taken; and save a muted chorus effect on "I'll Cry Alone" (the only remotely futuristic tune here), everything is translated with ultimate fealty. There's a distinct charm in that, one which makes the contents hummable to the point of occlusion of other thoughts. R.S.

BROTHER BAIT: ***Brother Bait.*** [Charles Green, producer; Phil Benton, engineer; recorded at Webb 4, Atlanta, Ga.] Tiger Lily TL 14070.

Performance: **Vibrant**
Recording: **Tight (in the good sense)**

With this, their debut album, Brother Bait has started off in high gear to say the least. Despite the unimaginative title, Bait has released an album of highly imaginative material. The material, though powerful, is not lacking in commercial appeal. However, I'm sure this band could be less commercial and still have plenty of appeal (only time will tell). The band, though from Atlanta, Ga., plays nowhere near the "Southern-Country-Rock" sound associated with that area (i.e. Allmans and Lynard Skynard). The sound is more English Rock; lots of vocal harmony, classical overtones on piano and an overall clean, tight sound.

Engineering was carried out with noticeable precision. The drums were panned well with great stereo separation and the vocals were happily placed well up front so you can hear the words!! Lead guitarist, Ron Bloom, plays a custom made twin-neck guitar (6 and 12 string) designed to his specifications (at a cost of $1,400). His guitar rides, cut through nicely with a perfect combination of guts and clearness. Bloom is also responsible for

writing all the material found on this recording. Possibly the only fault with the recording is that the bass doesn't seem to have as much bottom as one might expect from this type of band. It is a bit too high end (for my taste, anyway). However, it is full of punch, and bassist, Larry McDonald, definitely leaves no holes to be filled.

For those of you bored with bands hung up in the four-chord syndrome, an investment in Brother Bait's first album may prove to be a welcomed relief from the headaches caused from straining to hear the melody. C.F.-K.

BONNIE RAITT: *Sweet Forgiveness.* [Paul Rothchild, producer; John Haeny, Roger Mayer, engineers; recorded at Sunset Sound Recorders and Electra Sound Recorders, Los Angeles, Ca.] Warner Bros. BS 2990.

Performance: **Still in need of good material**
Recording: **How compressed can you get?**

This is Bonnie's sixth album, and I have yet to hear one that tracked 10 for 10. You'd expect someone who's recorded for as long as she has to find some worthwhile material. Still relying on the ideas and basic creativity of others, Raitt's and/or Rothchild's interpretations need work.

BONNIE RAITT: Relying on others

Knob wise, the major complaint with the performance turned in by the engineers is the constant use of compression and limiting. Overall, she sounds lifeless and bored throughout the album. The separations are good, but there's little excitement. The instruments keep their distance from the vocals, though the string effect in "Two Lives" is lost being so far back in the mix, and her hissing sibilants drive you up a wall. There is almost an overabundance of background vocals in group context. There isn't much musicianship in the grooves, either.

The title track suffers from the distorting guitar battling the acoustic piano for dominance. The drums throughout tend to be flat sounding, though the acoustic guitar is nice and crisp.

The disappointment lies with Raitt, although I come away with the feeling that this was an album made under duress of some sort. The fact that people keep going further and further back among the oldies for new material bothers me. When junk like Del Shannon's "Runaway" gets the nod, I begin to get nervous. Anyone for "Palisades Park?" G.P.

JAZZ

NOEL POINTER: *Phantazia.* [Dave Grusin and Larry Rosen, producers; Larry Rosen, engineer; recorded at Camp Columby, New City, New York.] Blue Note BNLA 736 H.

Performance: **Vital and electrifying**
Recording: **Uses all the gimmicks, but uses them well**

New York's High School of Music and Art has produced many jazz and pop musicians who've gone on to star-

dom, such as Billy Cobham and Jeremy Steig. The latest is an exciting contemporary violinist who seems more adept than most at fusing the jazz and rock idioms. Noel Pointer plays both acoustic and electric violin at different points on this recording, yet, even when he's amplified, he manages to avoid losing the characteristic string tone that makes a violin a violin and sounding like a substitute for a saxophone or lead guitar. Noel Pointer is a fiddler first and foremost. Through a varied repertoire that includes Stevie Wonder's "Living For The City," two jazz originals by guitarist Earl Klugh, a new arrangement of "Wayfaring Stranger" and the title song from "Fiddler On The Roof," Noel Pointer never loses the unique sound of his instrument. Of course, it helps to have players like Dave Grusin, Steve Gadd and John Tropea to work with. Also noteworthy is a flute player named Dave Valentin, who also wrote one of the songs on the album, "Rainstorm."

This certainly wasn't the kind of recording session where the engineer just opened the mics and let the guys blow. Bobby Columby of Blood, Sweat and Tears certainly knows the techniques that go into making a contemporary jazz/rock recording and he's capable of using them all at his studio in New City. As far as the overdubbing goes, some string and reed ensembles were added later in California but it still comes out a tasteful, integrated sound.

With his debut album Noel Pointer has served the public notice that he's here—and that he's a talent to be reckoned with. J.K.

MICHAEL HOWELL: *Alone.* [Pat Britt, producer; Mel Ford, engineer; recorded at Perceptive Studios, San Francisco, California, on August 17, 1976.] Catalyst CAT 7615.

Performance: **Brave virtuosity**
Recording: **Soft and fluffy**

In these days of electrified instruments it isn't easy to remember the time when guitars didn't have amplifiers. The acoustic Spanish guitar is an instrument nearly unmatched in its ability to communicate romantic beauty, warmth and melody. It's also amazingly self-sufficient, able to supply melody lines, bass lines and chordal harmonies. Concert guitarists such as Andres Segovia and Julian Bream have long been aware of the beauties of the unaccompanied guitar, as have such jazz greats as Eddie Lang and Joe Pass. Now they are joined by Michael Howell, who plays an ambitious program which includes McCoy Tyner's "Sama Layuca" and Duke Ellington's "Sophisticated Lady," in addition to his own "Honey Chile," Harvey

NOEL POINTER: A talent to be reckoned with

Mann's "Song No. 3" and Stevie Wonder's "Creepin'."

It would seem easy to record such a recital by a single instrument, but, when instructions such as no overdubbing or post-recorded sweetening are imposed, it becomes more difficult. A close miking would have brought more presence and detail, but it would have brought more extraneous finger noise as well. It was a wise decision on the part of the engineer and producer to go for a diaphanous sound which is admirably suited to the music and the artist.

Howell's decisions were not always as wise. After admitting, in the liner notes, the impossibility of duplicating McCoy Tyner's piano technique on guitar he goes for the impossible. It's called taking chances. Sometimes it works, sometimes it doesn't. But, when it does work, the results justify the times that Howell gets himself too far out on a musical limb for comfort. J.K.

THE GARY BURTON QUARTET WITH EBERHARD WEBER: *Passengers.* [Manfred Eicher, producer; Jan Erik Kongshaug, engineer; recorded at Talent Studio, Oslo, Norway.] ECM 1092.

Performance: **Spontaneous**
Recording: **Distinctive**

As is the case with practically all ECM recordings, each is recorded "live" in the studio, in one or two takes, with the actual recording completed in a day.

GARY BURTON: Always spontaneous and fresh

Add mixing time and the album is done in a week. Manfred Eicher believes that is the only way to record jazz and I agree. In the case of Gary Burton the results are always spontaneous and fresh.

The separations on *Passengers* are very distinctive, with Burton's shimmering vibraphone given the full spectrum of the mix. The concept of movement is interesting here, for while the vibraphone is basically the center of the mix, when he plays into the upper register the vibes are panned slightly to the right of the mix, and to the left for the lower octaves. Pat Matheny's guitar comprises the left of the mix along with Steve Swallow's bass, whereas Eberhard Weber's bass is to the right of the mix with the drums to the center. Drummer Gottlieb utilized his cymbals predominantly throughout this recording, which adds an odd tonal color when played against the vibes, for the resonances of the instruments are in contrast. Matheny's guitar is very cleanly recorded, with the occasional use of echo the only additive, which is a wise choice for a distortive-type effect would only muddy the overall performance.

My only problem with the recording was differentiating the two bassists, Swallow and Weber. I don't really feel that the use of two bassists added anything to the recording, with Swallow tending to be placed very much in the back of the mix. It was only through my knowledge that Weber's one-of-a-kind five-string bass has a high C-string that enabled me to discern the soloist.

EBERHARD WEBER: One-of-a-kind but hard to find

Common sense also told me that since Weber's the featured guest soloist, that must be his solo. Nevertheless, this is a very pretty jazz album filled with tonal colorations and combinations rarely employed today. G.P.

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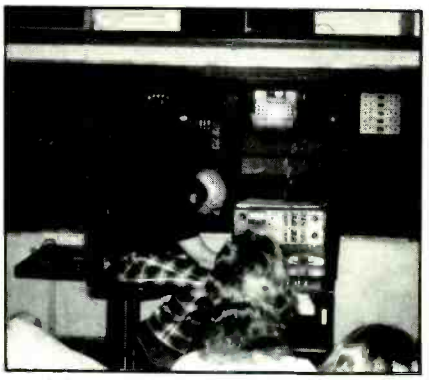

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CLASSICAL

TCHAIKOVSKY: *1812 Overture; Nutcracker Suite.* [Jack Kraft, keyboards; Larry Alexander, record engineer; Kraft & Alexander, synthesizer programming; recorded at Ultima Sound Studios, Blauvelt, N.Y.] London Phase-4 SPC 21168.

Performance: **Synthesized**
Recording: **Thin**

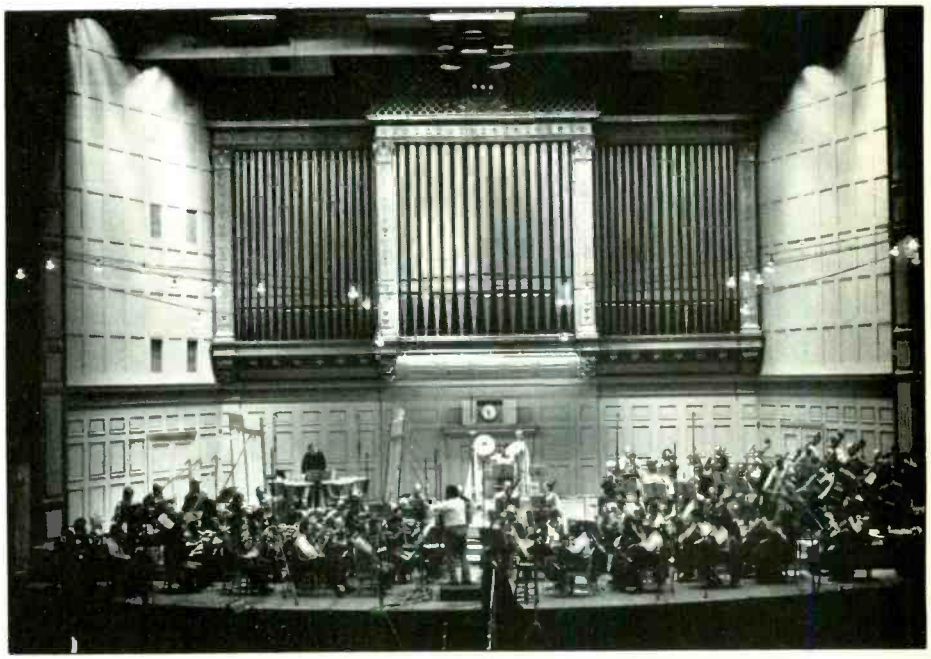
Figuring that I had spoken my peace about synthesizer versions of the classics in the April *MR*, I was surprised when I received this record for review. Maybe pop fans can get into this stuff, but I continue to receive negative vibes. The unsigned jacket notes state: "In the more densely orchestrated sections on this album, more than 200 individual tracks of synthesizer information were utilized to build a sound with a depth, power and variety of tonal textures equal to and greater than a full symphony orchestra." Nuts! Depth, power and tonal variety are precisely what one misses most of all! For the *1812 Overture*, listen to Haitink on Philips for an awesomely lifelike display of these three qualities (in the recorded sound, not the staid performance) or the Ormandy on RCA, which even utilizes an electronic cannon in the cataclysmic finale. In the *Nutcracker Suite*, I am surprised at the lack of dynamic variation and bass response. There's no substitute for the excitement of the real thing. The Arp 2600, Odyssey and Pro-Soloist synthesizers don't even come close. S.C.

SIBELIUS: *Symphony Nos. 1 and 2; Finlandia.* Boston Symphony Orchestra, Colin Davis, cond. [Vittorio Negri, producer; recorded in Symphony Hall, Boston, Mass.] Philips 9500.140 and 9500.141.

Performances: **First, dramatic; Second, relaxed**
Recordings: **Rich and clean**

The second and third installments of Colin Davis' Sibelius cycle with the Bostin Symphony Orchestra on Philips fully live up to the initial disc's coupling of the Fifth and Seventh Symphonies (*MR*, Dec/Jan 1976). The brass seem not as prominent, although it is difficult to judge whether the conducting or engineering is responsible. The winds are more closely balanced in the First, less so in the Second. The overall sound remains excellent, with a definite plus once again in Symphony Hall's superb acoustics.

The First is the most impressive performance. From the moody, sensitively phrased clarinet solo of the opening to the wrenching turmoil of the close, Davis makes the music surge and seethe with passion and drama. The well-judged wind balance reveals many felicitous details usually covered by the



COLIN DAVIS AND THE BOSTON SYMPHONY ORCHESTRA:
Strongly characterized interpretations

strings, and the harp contribution in the third movement is excitingly in focus. The only major drawbacks come in the otherwise superb final movement, where Davis holds back the first two statements of the motto theme, disrupting the line and destroying the built-in dramatic contrast with the concluding *largemente* treatment of the motto. Also, the broad tempo of the conclusion is not quite in the BSO's bones—the playing is not quite of one mind.

The Second Symphony may seem a bit casual in Davis' hands to those accustomed to the icily intense grandeur of Szell's classic Concertgebouw recording also on Philips. As with the First, Davis takes few liberties; tempo changes are very minor modifications of the basic pulse, with none of the passionate *accelerandos* or *rallentandos* of Bernstein or Barbirolli. But by the time Davis reaches the final three pages, he has imperceptively broadened into a tempo of extraordinary power, with the brass pealing out majestically over the solid underpinning of the glorious BSO strings.

Both recordings show occasional signs of haste: some less-than-sharp *ffz* attacks in the opening movement of the First, scrappy strings at one bar after letter F in the final movement, a sneeze just after the return of the allegro tempo at letter K, and a couple of bad splices; the Second contains a surprising number of almost-buried coughs. But these trifles should hardly discourage anyone from hearing Davis' strongly characterized interpretations. S.C.

SHOWS and SOUNDTRACKS

MIKLOS ROZSA: *Miklos Rozsa Conducts His Great Film Music.* [The Royal Philharmonic Orchestra, conducted by Miklos Rozsa; Brian Culverhouse, producer; Peter Bown, engineer; recorded in London, England.] Deutsche Grammophon 2584 013.

Performance: **Quite enjoyable**
Recording: **Fits the lushness of the music**

Miklos Rozsa has an enviable position among movie composers. He is both respected critically and popular commercially. Although his work spans roughly four decades of movie making, he was *the* Hollywood composer in the forties. The style and tone of movies in that period was his. As Christopher Palmer points out in the accompanying liner notes, Rozsa certainly does not have a light touch. There are times in movies like "The Lost Weekend" and "Spellbound" when the music overwhelms the films rather disastrously—making emotional transitions for actors, doing things the images should do, leaving nothing to the imagination.

In fairness to Rozsa, he did not dictate the style, he only embodied it most completely. Interestingly enough, this recording of Rozsa film music only has one cut from "Lost Weekend" to illustrate the original Rozsa sound. Every science fiction film for the next decade overworked the theremin after Rozsa established it as the evocation of emotional disturbance.

Fortunately, there is more to his music than that style. The first side of this DG release is all forties material except the cut from "A Time to Love and a Time to Die." "The Thief of Bagdad" music is brilliant and quite enjoyable, sounding like early William Walton. It is beautifully recorded as well, reproducing splendidly the glistening instrumental detail of the work. Notable, also, is the section from "Naked City," easily the most interesting music on the album and quite unlike most of Rozsa's music in its simplicity and clarity.

Side two turns to Rozsa, the epic composer. "Ben Hur" and "Quo Vadis," two notable Rozsa epic scores, are both regrettably absent here. The epic or spectacular film is a fine genre for Rozsa since the slightly overblown style of such films fits the lushness of Rozsa's music very well. Unfortunately, the selections here aren't very distinctive and tend to blur into one.

All in all, this is a most respectable disc, cleanly recorded with a full-balanced sound. Rozsa, the conductor, acquits himself well, indeed, showing great taste and restraint in his readings of his own music. The album is attractively packaged with notes by the ubiquitous Christopher Palmer and photos from the various films. I hesitate to point out that the photo from "A Double Life" labeled as Ronald Colman and Signe Hasso is not Ms. Hasso but a young Shelley Winters. A parting thought: there is much more Rozsa material of interest to collectors—"Double Indemnity," "The Asphalt Jungle" and "The Red House," just to name a few. How about a second Rozsa album, DG? H.R.

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MICROPHONES: SHure SM57CN $76.00; Shure SM58CN $91.00; E-V 1776 $83.00. CMG Sound, Inc. 12266 L Wilkins Ave., Rockville, Md. 20852 (301) 881-2660.

ENGINEER—I've sold my successful small studio in New England. I would like to work in a studio outside the U.S. or in Hawaii, the southwest or deep south. Some investment capital available in right circumstances. Pitt Kinsolving, 70 Turner Hill Rd., New Canaan, Ct. 06840 (203) 966-0765.

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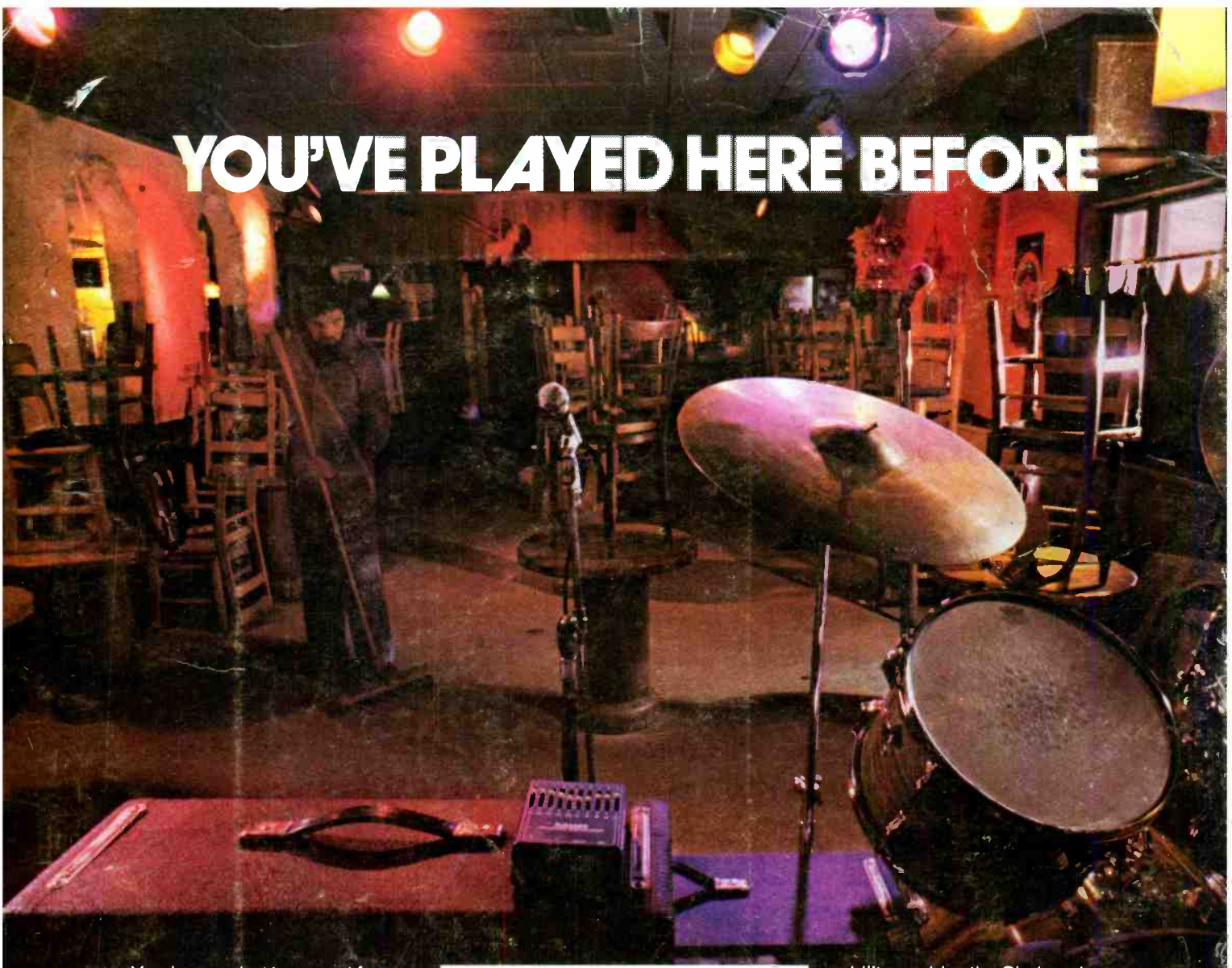

You know what to expect from brick walls, low ceilings and hard floors. Your bass gets eaten by the drapes and your guitar solo echoes right back at you. The environment of a small club is designed to make it a good place to drink and dance, not necessarily a good place to perform. What you need is the ability to tailor your sound to the room.

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